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MASTER THESIS

TITLE: Implementation of a Low Cost Video Wall using Raspberry Pi devices

MASTER DEGREE: Master of Science in Telecommunication Engineering & Management

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Thanks to Marc Fernández for guiding us during this project.
Thanks to i2CAT for providing us the necessary material.
Thanks to Toni Oller for supervising the project.

Overview

Keywords:
Video Wall, Raspberry Pi, H.264, Streaming

Description:

This project is a part of a more extensive project developed by i2CAT Foundation. The aim of this master thesis is to build a Low Cost Video Wall using Raspberry Pi devices.

A Video Wall can be defined as an array of monitors or screens synchronized to display content, simulating a bigger screen. This solution is the best alternative for cost-effective indoor and outdoor commercial applications where display images and video requires a larger screen size. A video wall solution can be composed of as many monitors or screens as necessary to cover a desired area.

On the other hand, the Raspberry Pi is a credit-card-sized single-board computer developed in the United Kingdom by the Raspberry Pi Foundation with the intention of promoting the teaching of computer science.

The most innovative fact about the Raspberry Pi is its price: for less than 30 euros it is possible to obtain a product that would have cost four or five times that amount before the Raspberry Pi arrived on the market. This means that large sums of money can be saved in some applications, especially in applications where a lot of separate devices are needed.

A Video Wall requires one device per screen, and therefore the implementation of a video wall using Raspberry Pi devices is a very interesting option. The main benefits are that energy consumption and cost are both reduced enormously, making video walls more accessible to enterprises which do not have massive funds, such as museums, schools, shops, galleries and offices.

To achieve that end, several assigments has been performed. First, a study of the operation of the video codecs supported by Raspberry Pi. Second, a testing of different software available to stream and visualize videos using the device, and finally, an analysis of the system to verify its correct behaviour.

Resum

Paraules clau:
Video Wall, Raspberry Pi, H.264, Streaming

Descripció:

Aquest projecte forma part d'un projecte més gran desenvolupat per la fundació i2CAT. L'objectiu d'aquesta tesi és construir un Video Wall de baix cost emprant els dispositius Raspberry Pi.

Podem definir un Video Wall com un conjunt de monitors o pantalles sincronitzades per mostrar contingut de vídeos o imatges, simulant una pantalla de major dimensió. Aquesta solució és una alternativa rendible per a les aplicacions comercials en interiors i a l'aire lliure on es requereix mostrar imatges o vídeos en una pantalla més gran. Una solució de Video Wall pot estar formada per tants monitors o pantalles com sigui necessari per cobrir l'àrea desitjada.

D'altra banda, el dispositiu Raspberry Pi és un ordinador de placa única de la mida d'una targeta de crèdit desenvolupada al Regne Unit per la Fundació Raspberry Pi, amb la intenció de promoure l'ensenyament de la informàtica.

El factor innovador de Raspberry Pi és el seu preu: per menys de 30 euros pots aconseguir un dispositiu amb unes característiques que haguessin costat quatre o cinc vegades aquesta quantitat abans de que aquest arribés al mercat. Això implica que es poden estalviar grans sumes de diners en algunes aplicacions, especialment en aquelles on es necessita una gran quantitat de dispositius independents.

Un Video Wall requereix un dispositiu per pantalla, i per tant l'implementació d'un Video wall utilitzant dispositius Raspberry Pi és una opció molt interessant. Els principals beneficis són que el consum d'energia i el cost es redueixen enormement, fent que els Video Walls siguin més accessibles per a les empreses que no compten amb grans pressupostos, com per exemple, museus, escoles, botigues, galeries i oficines.

Per tal d'aconseguir crear un Video Wall, s'han realitzat diverses tasques. En primer lloc, un estudi del funcionament dels còdecs de vídeo compatibles amb Raspberry Pi . En segon lloc, un test de diferents programes informàtics disponibles per fer streaming i visualitzar vídeos utilitzant aquest dispositiu i, finalment, un anàlisi del sistema per verificar el seu correcte funcionament.

INDEX

CHAPTER 1. INTRODUCTION 11

1.1. Work Structure 11

1.2. What is a Video Wall? 12

1.3. Project Objectives 14

1.3. Server Features 16
1.3.1. Server OS 16

1.4. Client Features 17
1.4.1. Client OS 19

CHAPTER 2. PROTOCOL ARQUITECTURE 20

2.1. Video Compression techniques 21

2.2. H.264 video codec 23
2.2.1. Profiles and levels 23
2.2.2. Structure 24

2.3. Real Time Protocol Operation 25
2.3.1 What is RTP 25
2.3.2. RTP functions 26
2.3.3. RTP packets and fields architecture 26

2.4. Real Time Control Protocol operation 29
2.4.1. What is RTCP 29
2.4.2. RTCP functions 29
2.4.3. RTCP fields and packets architecture 30

2.5. MPEG-2 Transmission 31
2.5.1. Building the MPEG Bit Stream 31
2.5.2. MPEG Transport Stream 32
2.5.3. Format of a Transport Stream Packet 33

CHAPTER 3. FRAMEWORKS 35

3.1. GStreamer 35

3.2. FFmpeg 37

3.3. VLC 39

3.4. OMXPlayer 40

CHAPTER 4. MILESTONES 42

4.1. Deployment scenario 1 42
4.1.1. Milestone 1 - Playing back a H.264 Full HD Video 43

4.2. Deployment scenario 2 45

4.2.1. Milestone 2 - Sending a HD/Full HD video to one Raspberry Pi 46
4.2.2. Milestone 3 - Sending a real time web cam video flow to one Raspberry Pi 50
4.2.3. Milestone 4 - Receiving a Full HD video / WebCam and playing back 51
4.2.4. Milestone 5 - Sending a Full HD video using MPEG-2 TS 52
4.2.5. Milestone 6 - Receiving a Full HD video using MPEG2-TS 53

4.3. Deployment scenario 3 54
4.3.1. Milestone 7 - Extracting and sending the audio stream from a Full HD video 55
4.3.2. Milestone 8 - Receiving and playing back the Full HD audio stream 55

4.4. Deployment scenario 4 56
4.4.1. Milestone 9 - Splitting and sending Full HD video to Raspberry Pi devices 57
4.4.2. Milestone 10 - Receiving a Full HD/Cam and playing back / cropping the video 60
4.4.3. Milestone 11 - Combining Milestone 2 and Milestone 7 60

4.5. Translation of GStreamer Pipelines to C Code 61

CHAPTER 5. SYNCHRONIZATION 62

5.1. GStreamer 62

5.2. FFmpeg 66

CHAPTER 6. CPU USAGE COMPARISON 67

6.1. GStreamer 67

6.2. FFmpeg 69

6.3. VLC 71

6.4. OMXPlayer 73

CHAPTER 7. NETWORK TESTING 74

7.1 DummyNet 74

7.2. DummyNet Scenario 75

7.3. DummyNet Configuration 75

7.4. Performed test and results 76

CHAPTER 8. THROUGHPUT AND BACKPLANE ANALYSIS 78

8.1. Milestone 9 - Throughput and Backplane 78

8.2. Milestone 11 - Throughput and Backplane 79

CHAPTER 9. VIDEO WALL CONTROL SYSTEM 81

9.1. Control system description 81

9.2. Startup scripts configuration 83

CHAPTER 10. COMPARISON WITH COMERCIAL PRODUCTS 85

10.1. SAGE Overview 85

10.2. SAGE Features and Tiled Display Wall Configurations 86

10.3. SAGE Minimum Systems Requirements and Minimun Budget 86

10.4. Minimum Systems Requirements and Budget of this project 88

10.5. Comparison of the power consumption results 89

10.6. Throughput and Backplane analysis 90

CHAPTER 11. PROJECT PLANNING 91

CHAPTER 12. CONCLUSIONS 93

CHAPTER 13. BIBLIOGRAPHY 96

ANNEXES 102

ANNEX A. RASPBIAN OS INSTALLATION 103
A1. Preparing our SD card for the Raspberry Pi 104
A2. Installation Raspbian “wheezy” 105
A3. Configuration of Raspberry Pi Network 106

ANNEX B. GSTREAMER INSTALLATION 111
B1. Installing GStreamer 1.0 in the Server 112
B2. Installing GStreamer 1.0 in the Clients 114

ANNEX C. VLC INSTALLATION 115
C1. Installing VLC in the Server 116
C2. Installing VLC in the Clients 116

ANNEX D. H261 AND H263 VIDEO CODECS 119
D1. H.261 video codec 120
D2. H.263 video codec 122

ANNEX E. GSTREAMER PIPELINES GRAPHS 123
E1. Milestone 1 125
E2. Milestone 2 125
E3. Milestone 3 125
E4. Milestone 4 125
E5. Milestone 5 126
E6. Milestone 6 126
E7. Milestone 9 126

Figure list

Figure 1. 3x3 Video Wall 12
Figure 2. System components 14
Figure 3. HP Pavilion DV6 Notebook PC 16
Figure 4. Debian OS Logotype 16
Figure 5. Raspberry Pi Model B components 17
Figure 6. Raspberry Pi Components Description 17
Figure 7. Raspbian OS 19
Figure 8. Set of protocols and video codecs available for the project 20
Figure 9. Protocol Layers involved in the Streaming 20
Figure 10. Block diagram of video compression 22
Figure 11. Motion compensation 22
Figure 12. RTP Header format 26
Figure 13. Combining elementary streams from encoders into a TS or a PS 32
Figure 14. Single Program Transport Stream (Audio and Video PES) 32
Figure 15. MPEG-2 Transport Stream Header Description 33
Figure 16. GStreamer framework logo 35
Figure 17. FFmpeg logo 37
Figure 18. VLC Logotype 39
Figure 19. Deployment scenario 1 42
Figure 20. Deployment scenario 2 45
Figure 21. FFmpeg statistics Milestone 1 47
Figure 22. FFmpeg statistics Full HD video 48
Figure 23. FFmpeg statistics (CRF applied) 49
Figure 24. Mark undefined packets – VLC 49
Figure 25. Deployment scenario 3 54
Figure 26. Deployment Scenario 4 56
Figure 27. Video cropping example 58
Figure 28. FFmpeg statistics 59
Figure 29: Video Wall syncronization example 62
Figure 30. GStreamer System clock and audio clock 62
Figure 31. GStreamer SPS, PPS, SEI and IDR frames 63
Figure 32. GStreamer non-IDR frames 63
Figure 33. H.264 bitmap of IDR frame 63
Figure 34. H.264 bitmap of non-IDR frames (P) 64
Figure 35. H.264 bitmap of non-IDR frames (B) 64
Figure 36. FU-A Marked packet 65
Figure 37. H.264 video flows packets 65
Figure 38. FFmpeg Wireshark Capture 66
Figure 39. Server CPU Usage Comparison 67
Figure 40. Raspberry 1 CPU Server Usage Comparison 68
Figure 41. Raspberry 2 CPU Server Usage Comparison 68
Figure 42. CPU Usage FFmpeg Server 69
Figure 43. CPU Usage FFmpeg Raspberry 1 70
Figure 44. CPU Usage FFmpeg Raspberry 2 70
Figure 45. VLC Server CPU Usage 71
Figure 46. VLC CPU usage Client 1 71
Figure 47. VLC CPU Usage Client 2 72
Figure 48. OMXPlayer CPU usage 73
Figure 49. DummyNet Operation 74
Figure 50. DummyNet Testing Scenario 75
Figure 51. Throughput 960x540 single flow 78
Figure 52. Backplane capacity Milestone 9 79
Figure 53. Throughtput 1920x1080 single flow 79

Figure 54. Backplane capacity Milestone 11 80
Figure 55. Commercial use scenario 81
Figure 56. Start up screen Raspberry Pi devices 82
Figure 57. Server VideoWall GUI 82
Figure 58. SAGE working demonstration 85
Figure 59. Project time line 92

Annex Figure List

Figure A1. SD Formatter V4.0 Configuration 104
Figure A2. Win32 Disk Imager Configuration 105
Figure A3. Advanced IP Scanner Program 106
Figure A4. PuTTY Configuration (1) 107
Figure A5. PuTTY Configuration (2) 107
Figure A6. Connection established 108
Figure A7. Session terminal of Raspberry Pi 108
Figure A8.Wireless Connection shared - Windows 7 109
Figure A9. Network interface configuration 109
Figure A10. DNS configuration file 110

Figure B1. A block diagram of H.261 video codec 120

Table list

Table 1. List of components and its functionality 15
Table 2. Raspberry Pi Minimum Recommended Specifications 18
Table 3. Frameworks and components relationship 35
Table 4. Initial Scripts in Raspberry Startup 81
Table 5. Initial Script for Audio 82
Table 6. Scripts in the server 82
Table 7. IP and Port configuration 83
Table 8. Basic SAGE Cost 87
Table 9. Our system cost 88
Table 10. 2x2 Project Video Wall cost 88
Table 11. Fundamental elements to compare with SAGE 88
Table 12. Basic SAGE Power Consumption 89
Table 13. Basic Power Consumption of our system 89
Table 14. Task table 92
Table 15. Frameworks results comparison 94
Table 16. H261 picture format 120
Table 17. Macroblocks in GOB - H261 120
Table 18. Luminance and Chrominance per macroblock 121
Table 19. H263 picture formats 122

CHAPTER 1. INTRODUCTION

This chapter focuses on explaining in detail the work structure of this project, what a video wall is, the objectives of the project and the main features of the components employed in this thesis.

1.1. Work Structure

The work structure of this project has been organized in 12 chapters. For each one, a brief description of the work done is provided.

In chapter 1, a description of the work structure used in this project is shown, a description of a video wall is explained, the project objectives are presented and the main system components needed are widely detailed.

In chapter 2, the video compression techniques have been studied, focusing on H264 video codec. This video codec provides enhanced compression efficiency and network friendly video representation for interactive and non-interactive applications (broadcast, streaming, storage and video on demand). It is designed to support a wide variety of applications and to operate over several types of networks and systems.

In chapter 3, an investigation of several software/frameworks available to form our video wall have been done. The frameworks are GStreamer, FFmpeg, VLC and OMXPlayer. They allow us to write a variety of instructions that permit to crop, stream and play back Full HD videos in our Raspberry Pi devices.

Chapter 4 is the core of the project and it is organized as a set of different milestones or test cases. It contains the deployment scenarios, the development of each milestone and the results obtained using different frameworks, previously explained in chapter 3.

Chapter 5 is focused on demostration of synchronization between the Raspberry Pi devices that set up the video wall. Different measures have been done using a network protocol analyser, called Wireshark, to evaluate the synchonization obtained using GStreamer and FFmpeg framework.

Chapter 6 is devoted to measure CPU usage in all the components of the videowall for every test case described in chapter 4. This provides information about which one of the frameworks is the most effective in terms of CPU usage.

Chapter 7 is devoted to test the system network. To perform the testing, a program called DummyNet has been used. It allows to know the behaviour of the video wall in a non-ideal network, by measuring the response of the system against network losses and delays.

In chapter 8, throughput and backplane analysis are performed. By using Wireshark, some measures are performed in the deployment scenarios to know the traffic that network components (router or switch) should support.

Chapter 9 describes how to implement a user friendly real case commercial scenario. It is a control system that allows the user to select the type of data to stream and how visualise it.

Chapter 10 provides a comparison of the system implemented with current commercial products (SAGE) in terms of cost, power consumption and backplane to demostrate that the system can compete in these three aspects.

Chapter 11 shows the timeline of the project and how tasks and time have been organized.

Finally, chapter 12 contains the project conclusions, a comparison of the results obtained for each framework and the environmental impact.

1.2. What is a Video Wall?

A video wall consists of multiple computer monitors, video projectors, or television sets tiled together contiguously or overlapped in order to form one large screen. Typical display technologies include LCD panels, LED arrays and rear projection screens.

Screens specifically designed for use in video walls usually have narrow bezels in order to minimise mullion - the gap between active display areas - and are built with long-term serviceability in mind. Such screens often contain the hardware necessary to stack similar screens together, along with connections to daisy chain power, video, and command signals between screens. A command signal may, for example, power all screens in the video wall on or off, or calibrate the brightness of a single screen after bulb replacement.

Reasons for using a video wall instead of a single large screen can include the ability to customize tile layouts, greater screen area per unit cost, and greater pixel density per unit cost, due to the economics of manufacturing single screens which are unusual in shape, size, or resolution.

Figure 1. 3x3 Video Wall

Video walls are sometimes found in control rooms, stadiums, and other large public venues. However,it is also possible to find simple video walls that can be driven from just using a set of personal computers and a set of monitors screen at home.

Video Wall Latest Advancements

Researchers around the U.S. have been advancing video wall technology in recent years and are listed below:

SAGE

Jason Leigh and others at the Electronic Visualization Laboratory, University of Illinois, Chicago, developed SAGE, the Scalable Adaptive Graphics Environment, allowing the seamless display of various networked applications over a large display wall (LDW) system.

Different visualization applications such as 3D rendering, remote desktop, video streams, and 2D maps, stream their rendered pixels to a virtual high-resolution frame buffer on the LDW. Using a high-bandwidth network, remote visualization applications can feed the streams of the data into SAGE.

The user interface of SAGE, which works as a separate display node, allows users to relocate and resize the visualization stream in a form of window, which can be found in a conventional graphical user interface. Depending on the location and size of the visualization stream window on the LDW, SAGE reroutes the stream to respective display nodes.

Chromium

Chromium is an OpenGL system for interactive rendering on graphics clusters. By providing a modified OpenGL library, Chromium can run OpenGL-based applications on a LDW with minimal or no changes. One clear advantage of Chromium is utilizing each rendering cluster and achieving high resolution visualization over a LDW.

Chromium streams OpenGL commands from the application node to other display nodes of a LDW. The modified OpenGL library in system handles transferring OpenGL commands to necessary nodes based on their viewport and tile coordinates.

Media Fusion

David Hughes and others from SGI developed Media Fusion, an architecture designed to exploit the potential of a scalable shared memory and manage multiple visual streams of pixel data into 3D environments. It provides data management solution and interaction in immersive visualization environments. Its focus is streaming pixels across heterogeneous network over the Visual Area Network (VAN) similar to SAGE.

However, it is designed for a small number of large displays. Since it relies on a relatively small resolution for the display, pixel data can be streamed under the fundamental limit of the network bandwidth.

Hiperwall, Inc.

Based on video wall display technology developed at University of California, Irvine's California Institute for Telecommunications and Information Technology (Calit2), Hiperwall, Inc. was formed to commercialize a video wall system that offers advanced software functionality and interface with standard PCs, monitors and Ethernet network.

The system displays high-resolution still images, HD videos, live HD video streams and PC applications. Multiple feeds can be displayed on the wall simultaneously and users can reposition and resize each feed in much the same way they move and resize windows on a PC desktop. Each feed can be scaled up for viewing on several monitors or the entire wall instantly depending upon the user's discretion.

1.3. Project Objectives

The requirements that should be complied in the implementation of our Low Cost video Wall are the following:

1. Low Cost
2. Being able to play back a Full HD 1080p video stream.
3. Providing synchronization between Raspberry Pi's.
4. Using H.264 as a video compression format.
5. Use of Real Time Protocol as Application layer.
6. Use of User Datagram Protocol as Transport Layer.
7. Minimum latency in reception.
8. Scalable.The aim is to increase easily the screen matrix (2x2, 3x3, 4x4, etc.).
9. Robust. The system can be recovered in case of network losses or network delays.

To implement this, a notebook that will act as a server will be used to emit and encode the video streams and two Raspberry Pi devices to decode and visualize each part of the video. The criteria to use Raspberry Pi devices is that they provide H.264 hardware decoding[43] and that they are very cheap.

The following figure shows the components of our video wall system. Later on, in the following sections, the components utilized in this project are explained in detail.

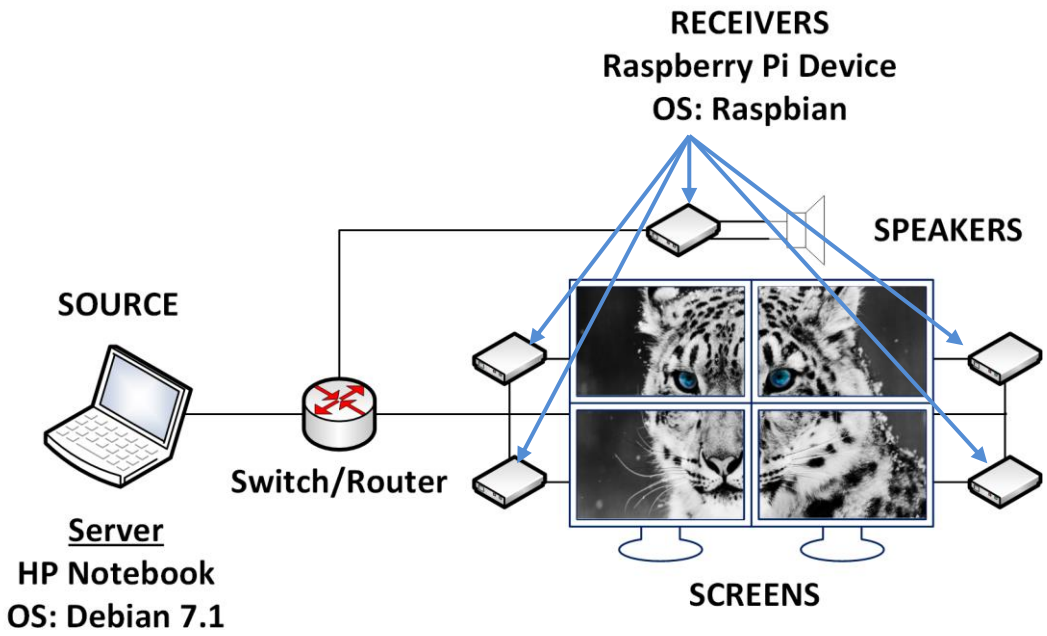


Figure 2. System components

Component	Description and Functions
Source	The source is a notebook that acts as a server and its function is to stream encoded audio and video flows to the receivers using some scripts.
Switch / Router	This element is in charge of distributing the flow sent by the server to the different receivers.
Receivers	The receivers are the five Raspberry Pi devices. Four of them are running a script that decode, crop and play back the video stream. The fifth Raspberry is responsible for decoding the audio stream and play back the audio through the speakers.
Speakers	The speakers are connected to the Raspberry Pi through a 3.5 mm jack to provide stereo audio.
Screens	Screens are connected to Raspberry Pi devices using HDMI to HDMI lead or HDMI to DVI lead.

Table 1. List of components and its functionality

1.3. Server Features

The server in this project is a HP Pavilion DV6 Notebook PC.

Figure 3. HP Pavilion DV6 Notebook PC

The server has the following features:

- **CPU:** Intel Core i3 M330 2.13 GHz (4 CPUs)
- **RAM:** 4 GB
- **GPU Card:** NVIDIA GeForce G105M
- **Operating System (OS):** Debian GNU/Linux 7.1 (wheezy)
- **Hard Disk Drive Size:** 78 GB Logic Partition

1.3.1. Server OS

The server uses Debian GNU/ Linux 7.1 as OS because Raspberry has installed a version of Debian for Raspberry Pi devices. Furthermore all the programs used during the project are compatible with Debian.

Debian provides more than a pure OS: it comes with over 37500 packages; precompiled software bundled up in a nice format for easy installation on the laptop.

Figure 4. Debian OS Logotype

The Debian Project is an association of individuals who have made common cause to create a free operating system.

1.4. Client Features

The Raspberry Pi is a credit-card sized computer that plugs into the TV and a keyboard. It is a capable little PC which can be used for many of the things that a desktop PC does, like spreadsheets, word-processing and games. The more important thing for this project is that it can play high-definition video.

During the project 2 Raspberry Pi (Model B) have been used. Its components are shown in Figure 5.

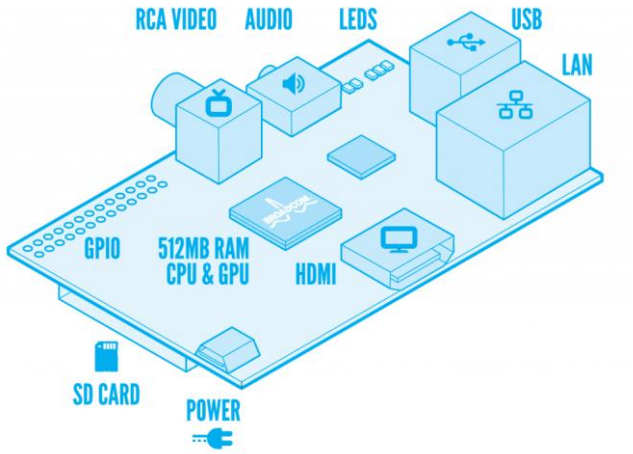


Figure 5. Raspberry Pi Model B components

Figure 6 and Table 1 show the minimum recommended specifications for each component.

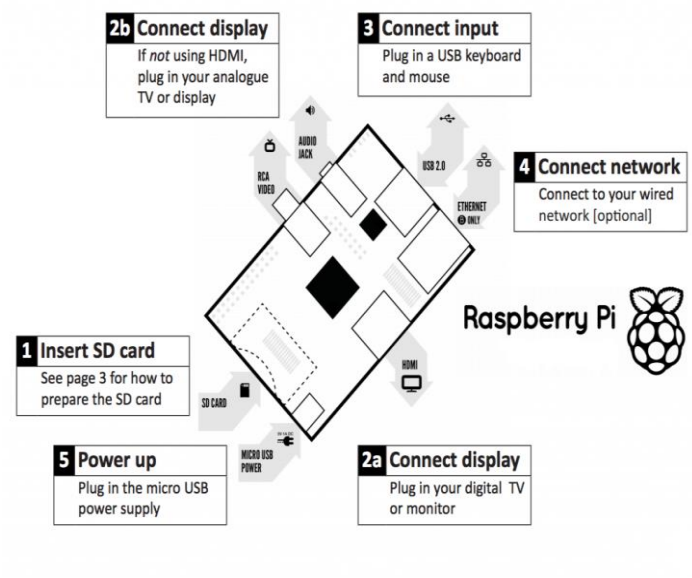

2b Connect display
If not using HDMI, plug in your analogue TV or display
3 Connect input
Plug in a USB keyboard and mouse
4 Connect network
Connect to your wired network [optional]
1 Insert SD card
See page 3 for how to prepare the SD card
5 Power up
Plug in the micro USB power supply
2a Connect display
Plug in your digital TV or monitor
Raspberry Pi

Figure 6. Raspberry Pi Components Description

Item	Minimum recommended specifications
1 - SD card	Minimum size 4Gb; class 4 (the class indicates how fast the card is). It is recommended using branded SD cards as they are more reliable.
2a - HDMI to HDMI / DVI lead	HDMI to HDMI lead (for HD TVs and monitors with HDMI input) or HDMI to DVI lead (for monitors with DVI input).
2b - RCA video lead	A standard RCA composite video lead to connect to your analogue display if you are not using the HDMI output.
3 - Keyboard and mouse	Any standard USB keyboard and mouse should work. Keyboards or mice that take a lot of power from the USB ports, however, may need a powered USB hub. This may include some wireless devices.
4 - Ethernet (network) cable [optional]	Networking is optional, although it makes updating and getting new software for the Raspberry Pi much easier.
5 - Power adapter	A good quality, micro USB power supply that can provide at least 700mA at 5V is essential. Many mobile phone chargers are suitable - the label on the plug needs to be checked. If the supply provides less than 5V then the Raspberry Pi may not work at all, or it may behave erratically. Be wary of very cheap chargers: some are not what they claim to be. It does not matter if the supply is rated at more than 700mA.
6 - Audio lead [optional]	If HDMI is used, digital audio will be achieved via this. If the analogue RCA connection is used, stereo audio will be available from the 3.5mm jack next to the RCA connector.

Table 2. Raspberry Pi Minimum Recommended Specifications

Focusing the attention on the model of 2 Raspberry Pi devices, these devices have the following features:

- **Size:** 85.60 mm × 53.98 mm
- **CPU:** ARMv6 –compatible processor rev 7 (v61) - (700 MHz)
- **GPU Card:** BroadCom BCM2708[1,2,3,4]
- **SDRAM:** 512 MB

- 2 x USB 2.0 , 10/100 MBIT
- HDMI output.

Some small differences that will affect the performance have been encountered between both Raspberry Pi devices. This is a constraint because the RAM of both Raspberry Pi devices are not the same:

Raspberry 1 (Son Goku)
- Samsung 240 K4P4G324EB-AGC1 **GKI59790**

Raspberry 2 (Krilin)
- Samsung 240 K4P4G324EB-AGC1 **GKI597M**

It is not possible to know whether there are differences in the BroadCom SoC because they are physically placed under the Samsung RAM.

1.4.1. Client OS

Raspbian[5,6] is a free operating system based on Debian optimized for the Raspberry Pi hardware.

The initial build of over 35,000 Raspbian packages, optimized for best performance on the Raspberry Pi, was completed in June of 2012. However, Raspbian is still under active development with an emphasis on improving the stability and performance of as many Debian packages as possible.

Figure 7. Raspbian OS

Raspbian is an unofficial port of Debian Wheezy **armhf** with compilation settings adjusted to produce optimized "hard float" code that will run on the Raspberry Pi. This provides significantly faster performance for applications that make heavy use of floating point arithmetic operations.

All other applications will also gain some performance through the use of advanced instructions of the ARMv6 CPU in Raspberry Pi.

Although Raspbian is primarily the efforts of Mike Thompson and Peter Green, it has also benefited greatly from the enthusiastic support of Raspberry Pi community members who wish to get the maximum performance from their device.

In **Annex A**, how to install and configure Raspbian OS in Raspberry Pi devices is explained.

CHAPTER 2. PROTOCOL ARQUITECTURE

This chapter is centered on the different standards of real time video transmission. The different protocols used in the project to implement the video wall are explained.

The following two figures describe the set of protocols and video codecs used in the project. In the following sections the most relevant aspects of H.264, Real Time Protocol (RTP) and Real Time Control Protocol (RTCP) are explained.

In Figure 8, the set of protocols and video codecs available for implement this project are described. The protocols involved are: Internet Protocol (IP), User Datagram Protocol (UDP), RTP and RTCP. There is also a set of audio and video codecs available and the user interface defined in chapter 9.

<table>
<tr><td colspan="3">Multimedia Application – User Interface</td></tr>
<tr><td colspan="3">Media Control</td></tr>
<tr><td>Audio Codecs

G.711
G.723.1
G.729</td><td>Video Codecs

H.261
H.263
H.264</td><td rowspan="2">RTCP</td></tr>
<tr><td colspan="2">RTP (H.264 or MPEG2-TS container)</td></tr>
<tr><td colspan="3">UDP</td></tr>
<tr><td colspan="3">IP</td></tr>
</table>

Figure 8. Set of protocols and video codecs available for the project

Figure 9 shows the protocol layers involved in the streaming in both elements: the source (Server) and the receivers (Client). The protocol layers follow the standard of OSI stack model.

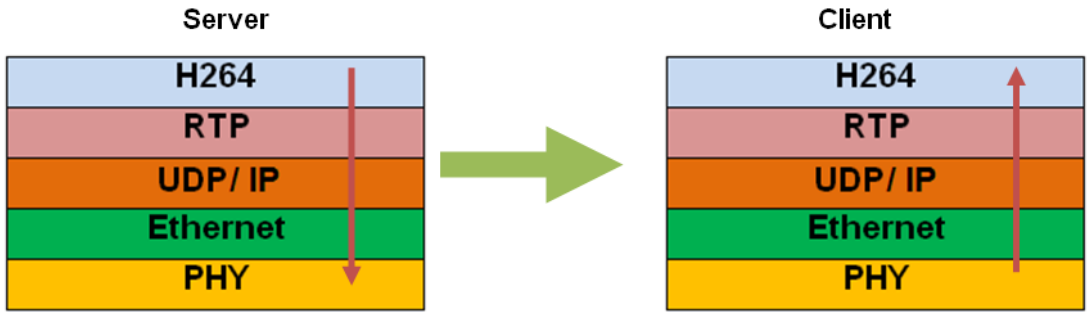


Figure 9. Protocol Layers involved in the Streaming

2.1. Video Compression techniques

Video applications require some form of data compression to achieve reasonable precondition for storage and transmission. The digital video compression is one of the main issues in digital video coding, enabling efficient distribution and interchange of visual information.

Video codecs are devices that are used to compress and decompress as well as to encode and decode video streams. The most complex part of a codec is the compress/decompress function. Codecs can do their work by hardware but also by software with fast processors. The main goal of coding is the bit-rate reduction for storage and transmission of the video source while retaining video quality as good as possible. There are a number of international standards and also many proprietary techniques for digital video compression. The basic idea behind video compression is to remove spatial redundancy within a video frame and temporal redundancy between adjacent video frames.

There are two main types of compression techniques, lossless and lossy. In the lossless compression a frame can be decompressed into the original exactly. The compression ratio of lossless methods (Huffman, Arithmetic, LZW, RLE) is not high enough for digital video communication. In the lossy compression compressed data that can be decompressed into images that look similar to the original (as human eye sees them) is created, though they are different in digital form.

The human eye is more sensitive to changes in brightness than to chromaticity changes. Therefore the image data is first divided into one luminance and two chrominance components, and the chrominance components are subsampled relative to the luminance component.

After this step the usual lossy compression method used in digital video compression is based on Discrete Cosine Transform (DCT) and quantization. This technique reduces the high spatial frequency components from the image since the human viewer is more sensitive to the reconstruction errors of low frequency components.

The purpose of the quantization step is to represent the DCT-coefficients with the precision what is needed to achieve the required image quality. The zig-zag step arranges the high frequency coefficients to the end of the stream and since most of them have become zero after the quantization, run length encoding (RLE) is used for further compression.

The upper left corner coefficient represents the mean value of the block and is encoded using the difference from the previous block (DPCM). The final step in the compression process is to minimize the entropy using Huffman or arithmetic coding. The encoded frame is often called I-frame (intra frame) because the encoding process uses no information from other frames. The block diagram of the encoding process is represented in Figure 10.

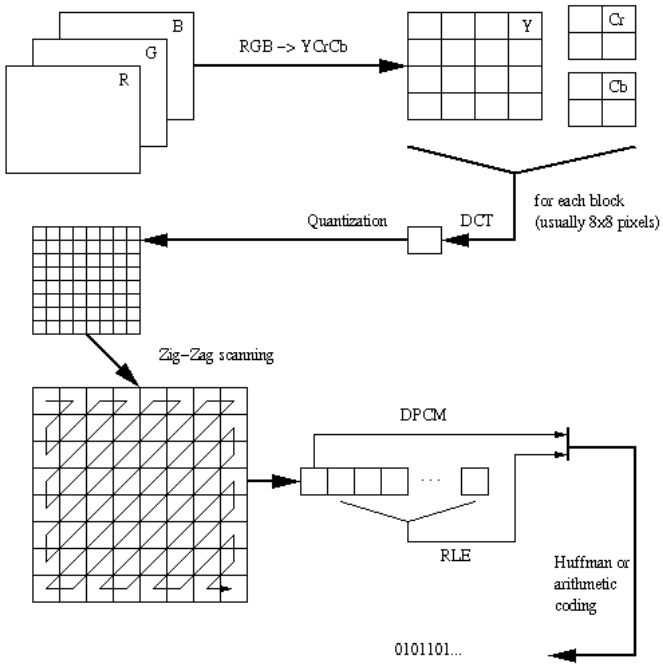


Figure 10. Block diagram of video compression

In addition to the previous compression technique, the temporal redundancy between frames can be utilized for further compression. The basic method is to calculate the prediction error between corresponding blocks in the current and previous frames. The error values are then sent to the compression process. Compressed frames generated using predictions are usually called P-frames. When using both previous and future frames as reference, the frame is called B-frame (bidirectional frame).

Motion compensated prediction is an efficient tool to reduce temporal redundancy between frames. The concept of motion compensation contains the motion estimation between video frames (Figure 11). The motion is described by a small number of motion vectors which gives the translation of a block of pixels between frames. The motion vectors and compressed prediction errors are then transmitted.

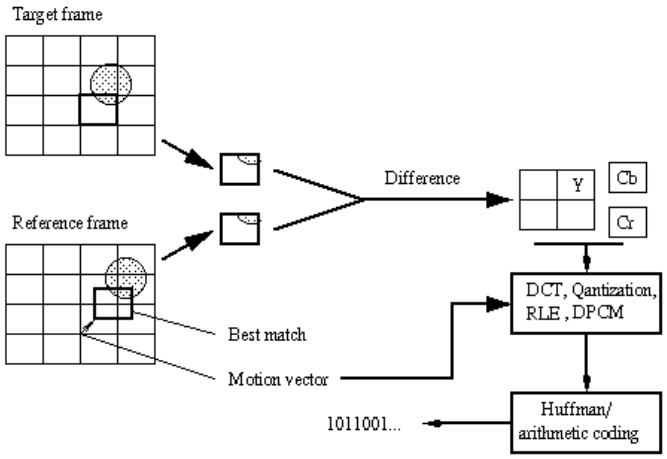


Figure 11. Motion compensation

2.2. H.264 video codec

H.264/AVC[8], also known as MPEG-4, is the latest international video coding standard. It was jointly developed by the ITU-T (International Telecommunication Union) “Video Coding Experts Group” (VCEG) and the ISO/IEC (International Organization for Standardization/ International Electrotechnical Commission) “Moving Picture Experts Group” (MPEG).

The importance of new network access technologies like cable modem, xDSL (Digital Subscriber Line) and UMTS created demand for the H.264/AVC[11] standard. For this reason, the goals of this standardization effort were enhanced compression efficiency and network friendly video representation for interactive (video telephony) and non-interactive applications (broadcast, streaming[10], storage and video on demand).

Therefore, H.264/AVC[14] was designed to support a wide variety of applications and to operate over several types of networks and systems. As a result, this standard provides gains in compression efficiency of up to 50 % over a wide range of bit rates and video resolutions compared to previous standards (H261, H263).

This video codec is used due to the project requeriments and, furthermore, it is available on the frameworks to encode the video, and in the Raspberry Pi, to decode the video.

2.2.1. Profiles and levels

The H.264/AVC standard defines profiles and levels in order to maximize the interoperability while limiting the complexity. A profile is defined as a subset of coding tools that can be used to generate a **bitstream**, whereas a level places constraints on certain key parameters of the bitstream.

Therefore, by means of profiles and levels, minimum bounds on the decoding capabilities can be set in order to target different application domains. Hence, H.264/AVC defines different profiles. The ones listed below are the most important:

High: Primary profile for broadcast and disc storage applications, particularly for high-definition television applications.

Baseline: Designed to minimize the complexity and provide high robustness and flexibility, this profile is widely used in video conferencing and mobile applications.

Main: This profile typically allows the best quality at the cost of higher complexity and delay.

Extended: This profile was designed to combine the robustness of the Baseline profile with a higher degree of coding efficiency and greater network robustness for such applications as flexible video streaming.

It is important to say that B slices are only allowed in the main profile and above. They can be used to save on bandwitdth but are harder to decode, which is why some devices might not support them.

All types of profiles have been tested, but the one used during the deployment of this project has been High profile because it is adopted for high definition applications such Blu-ray and DVB HDTV.

2.2.2. Structure

The H.264/AVC standard defines two conceptual layers: the “Video Coding Layer” (VCL) and the “Network Adaptation Layer” (NAL). The first one defines the efficient encoding representation of the video whereas the second one is designed to provide “network friendliness” by facilitating the ability to map H.264/AVC VCL data to different transport layers.

As in all prior ITU-T and ISO/IEC JTC1 (Joint Technical Committee 1) video standards since H.261, the VCL design follows the so-called “block-based hybrid video coding approach” in which each coded picture is divided into block-shaped units called “macroblocks”. Each one of them consists of three components: Y, Cr and Cb, where Y is the luminance component which represents brightness information and Cr and Cb are the chrominances and represent the colour information. Due to the fact that the human visual system is more sensitive to luminance than to chrominance, the chrominance signals are both subsampled by a factor of 2 in horizontal and vertical direction. Therefore, a macroblock consists of one block of 16x16 samples for the luminance component and two blocks of 8x8 samples for the chrominance. This is called 4:2:0 sampling with 8 bits of precision per sample.

These macroblocks are coded in Intra or Inter mode. Then, the prediction error, which is the difference between the original and the predicted block, is transformed using a 4x4 integer transformation with similar properties as the 4x4 DCT (Discrete Cosine Transform). The task of the transformation is to reduce the spatial redundancy of the prediction error signal. As a result of the transformation, a block of transformed coefficients is obtained which is next quantized using a “quantization parameter” (QP) that can take up to 52 different values (0-51) when video format supports 8 bits per decoded sample. Typically, the result is a block in which most or all the coefficients are zero, with few non-zero coefficients.

The aim of the quantization process is to reduce the precision of the coefficients according to the QP. If QP is set to a high value (the step size doubles with each 6 increments of QP), then more coefficients are set to 0 whereas if QP is set to a low value, then more non-zero coefficients will remain after the process. Thus, this process keeps the information from the low frequencies whereas the high frequencies are set to 0.

Next, the quantized transform coefficients of a block are generally scanned in a zig-zag fashion and then transmitted using entropy coding methods. In H.264/AVC, two methods of entropy coding are supported: a low complexity technique based on the usage of contextadaptively switched sets of variable length codes (Context-Adaptive Variable Length Coding (CAVLC)) and the algorithm of context-based adaptive binary arithmetic coding (Context- Adaptive Binary Arithmetic Coding (CABAC)).

In order to reconstruct the same image on the decoder side, the dequantized coefficients are inversed transformed and added to the prediction signal. The result is the reconstructed macroblock.

A sequence of macroblocks is grouped together forming a “slice”. Therefore, a picture is a collection of one or more slices in H.264/AVC. Moreover, slices are selfcontained in the sense that the data within them can be correctly decoded without use of data from other slices belonging to the same frame. As a consequence, in an IP network environment each slice corresponds to a packet sent through the network.

2.3. Real Time Protocol Operation

2.3.1 What is RTP

RTP[18] provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services; such as videoconferences. RTP does not address resource reservation and does not guarantee quality of service for real-time services.

The use of this protocol is justified to transport information that requires a special processing. This information must be available in real time for the receptor, and therefore, must be strictly controlled to permit a correct reception.

It must be clarified that in all communications latency between receptor and transmitter always exists. Depending on the applications, the requirements will be more or less restrictive although all are real time applications. For example, online video visualization through the Network is considerated a real time application because it uses video and audio flows. The same occurs in videoconference applications. The margins of second application are more restrictive.

In general and regarding real time applications, it is important to control the latency, jitter and packet order arrival, as well as losses that might come, to ensure the application proper functioning.

For all the reasons explained above, RTP fits perfectly with the project requirements to be used as an application layer.

It is interesting to remark that RTP uses User Datagram Protocol (UDP) as a transport layer. TCP (Transmission Control Protocol) is also a transport protocol that offers a reliable video or audio stream but it will slow down transmission and re-transmit corrupted packets in order to achieve that. On the other hand, UDP does not guarantee the reliability of the communication and, therefore, will not slow down or re-transmit data.

The application implemented in this project does not need a reliable data stream and, in case of generating corrupted packets or losing any of them, the application does not have a mechanism available for their retransmission. Therefore, UDP is a better choice.

RTP functionalities will be explained in the next section.

2.3.2. RTP functions

The RTP protocol carries out a control task on media data flow adding, for example, temporal marks and sequence numbers in each data packet to identify them in reception. Furthermore, the protocol is capable of identifying transported information according to the type (audio or video) and separate it in diverse flows. The principal functions that can be performed using the RTP protocol are:

Calculating delays in communication

Temporal marks (timestamp) identify when data was sent and when it was received.

Synchronizing diverse flows if necessary

The temporal marks allow executing this synchronization (for example, an audio flow associated to video).

Receiving the packets and processing in correct order

For this purpose, the sequence number is used. With it, it is possible to know which one is the next packet to deal with, rearrange it if necessary and calculate the losses in the system. In some cases, the sequence number is the only way to identify various packets that can transport the same temporal mark (if the packet has been divided).

Finally, although it is not a specific protocol function, it is important to mention that this does not contemplate the retransmission of data since there is no need to retransmit information that cannot be used.

2.3.3. RTP packets and fields architecture

Below these lines, the RTP header is shown. The minimum RTP header size is 12 bytes (the fields marked in blue).

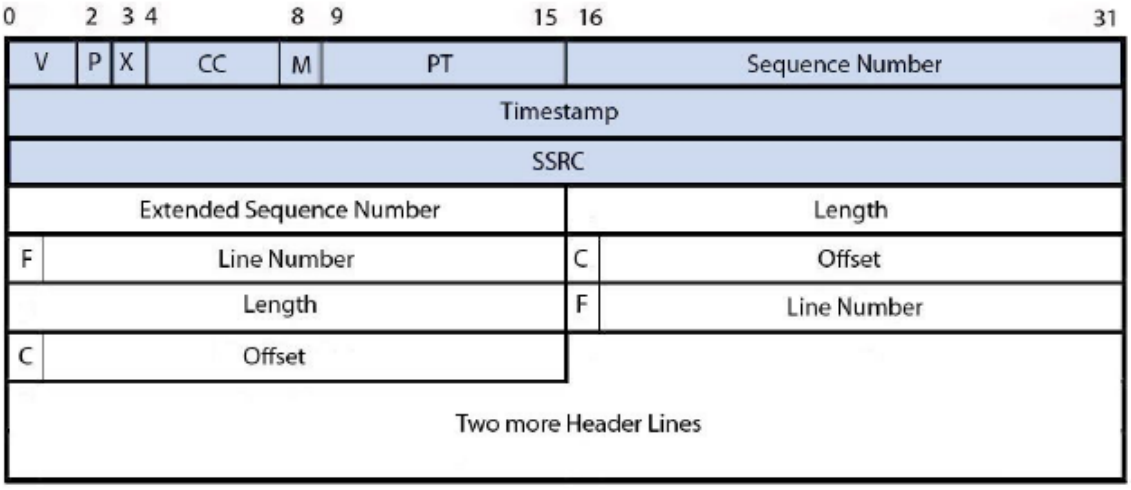


Figure 12. RTP Header format

The description of each of the fields in the header is the following:

Version Number (V) (2 bits)

Indicates the version of the protocol. The current version is 2.

Padding (P) (1 bit)

It is used to indicate if there are extra padding bytes at the end of the RTP packet.

Extension(X) (1 bit)

It indicates presence of an extension header between standard header and payload data.

CSRC Count (CC) (4 bits)

Contains the number of Contributing Source identifiers that follow the fixed header.

Marker (M) (1 bit)

Marker bit. If progressive scan video is being transmitted, the marker bit denotes the end of a video frame. If interlaced video is being transmitted, it denotes the end of the field. The marker bit must be set to 1 for the last packet of the video frame/field. It must be set to 0 for other packets.

Payload Type (PT) (7 bits)

A dynamically allocated payload type field.

Sequence Number (16 bits)

The low-order bits for RTP sequence number. The standard 16- bit sequence number is augmented with another 16 bits in the payload header in order avoid problems due to wrap-around when operating at high rate rates.

Time Stamp (32 bits)

For progressive scan video, the timestamp denotes the sampling instant of the frame to which the RTP packet belongs. Packets must not include data from multiple frames, and all packets belonging to the same frame must have the same timestamp.

Progressive scanning (alternatively referred to as noninterlaced scanning) is a way of displaying, storing, or transmitting moving images in which all the lines of each frame are drawn in sequence. This is in contrast to interlaced video used in traditional analog television systems where only the odd lines, then the even lines of each frame (each image called a video field) are drawn alternately. Progressive scanning is universally used in computing.

For interlaced video, the timestamp denotes the sampling instant of the field to which the RTP packet belongs. Packets must not include data from multiple fields, and all packets belonging to the same field must have the same timestamp.

SSRC (32 bits)

Synchronization source identifier uniquely identifies the source of a stream.

The optional header extensions are described below:

Extended Sequence Number (16 bits)

The high order bits of the extended 32-bit sequence number, in network byte order.

Line No (15 bits)

Scan line number of encapsulated data, in network byte order. Successive RTP packets may contain parts of the same scan line (with an incremented RTP sequence number, but the same timestamp), if it is necessary to fragment a line.

Offset (15 bits)

Offset of the first pixel of the payload data within the scan line. If YCbCr format data is being transported, this is the pixel offset of the luminance sample; if RGB format data is being transported, it is the pixel offset of the red sample; if BGR format data is being transported, it is the pixel offset of the blue sample. The value is in network byte order. The offset has a value of zero if the first sample in the payload corresponds to the start of the line, and increments by one for each pixel.

F (1 bit)

Field Identification identifies which field the scan line belongs to, for interlaced data. F=0 identifies the first field and F=1 the second field. For progressive scan data (e.g., SMPTE 296M format video), F must always be set to zero.

Length (16 bits)

Numbers of octets of data included from this scan line, in network byte order. This must be a multiple of the pgroup value.

C (1 bit)

Continuation determines if an additional scan line header follows the current scan line header in the RTP packet. Set to 1 if an additional header follows, implying that the RTP packet is carrying data for more than one scan line. Set to 0 otherwise.

Several scan lines may be included in a single packet, up to the path MTU limit. The only way to determine the number of scan lines included per packet is to parse the payload headers.

2.4. Real Time Control Protocol operation

This section aims to provide a global vision of RTCP operational (RFC 3550), their functions and the packet types contemplated.

RTCP[18] is introduced as an adjacent control protocol to the RTP. The RTCP functionality is to report the network state to different nodes of communication.

As it is explained below, RTCP is a sister protocol of RTP and also it has been used in this project during the streaming process to control and synchronize the audio and video flows.

2.4.1. What is RTCP

RTCP provides out-of-band control information for an RTP flow. It is a sister protocol of RTP and helps it in the delivery and packaging of multimedia data, but does not transport any data itself. It is used periodically to transmit control packets to participants in a streaming multimedia session. The primary function of RTCP is to provide feedback on the quality of service being provided by RTP.

RTCP gathers statistics on a media connection and information such as bytes sent, packets sent, lost packets, jitter, feedback and round trip delay. An application may use this information to increase the quality of service, perhaps by limiting flow or using a different codec.

This protocol does not offer itself any kind of response to any event on the network, but it is a simple informant. The application layer will be the responsible for carrying out the relevant actions.

2.4.2. RTCP functions

The RTCP protocol implements three different basic functionalities:

To provide information of transmission state and quality to the involve nodes

This is a totally necessary function for RTP transport protocol, to be able to implement the flow control and congestion. Receiver and Sender Reports provide this information.

Implementation able to sync to various flows

This solves the problem that involves the use of the SSRC from the RTP protocol. This field identifies every RTP flow avoiding confusion in the transmission. The same source can define additional different flows (for example audio and video) with different SSRC. Order to properly synchronize, RTCP use a new field called CNAME (canonical name) identifying a single emitter. Can be associated diverse SSRC into same CNAME.

To realize the two previous functions, all RTP session nodes send RTCP packets. The send tax of these packets should vary depending on number of participants in the session, for optimal use of the network.

Session Control Information

Finally, RTCP can carry session control information. For example, it could carry information regarding the participant identification and displayed to the user. The first two functions must be met in all possible RTCP scenarios, but are especially important in multicast sessions. The sending of RTCP packets should be controlled independently from the transmitter and receiver to avoid critical situations in unidirectional channels where dialogue is impossible for the nodes.

2.4.3. RTCP fields and packets architecture

RTCP packet types defined according to their function and control information they carry. Below, there is a brief summary:

Sender Report (SR)

This sends statistics information for receipt and transmission. Active transmitters send them.

Receiver Report (RR)

This sends reception information statistics. No transmitters send them.

Source Description (SDES)

Identifies source, including CNAME.

End of participation (BYE)

It is a field that indicates the source disconnection.

Application-specific message (APP)

It indicates proper application functions.

All RTCP packets have a fixed part and other variable part that depending on the type, have different length. This length must be a multiple of 32 bits. This is important because several RTCP packets can be concatenated without any kind of separation to form a bundle RTCP.

2.5. MPEG-2 Transmission

The MPEG-2[15] (Moving Picture Expert Group-2) standards define how to format the various component parts of a multimedia program (which may consist of: MPEG-2 compressed video, compressed audio, control data and/or user data). It also defines how these components are combined into a single synchronous transmission bit stream. The process of combining the streams is known as *multiplexing*.

MPEG-2 has been used in this project to stream and receive correctly video flows using a framework that only it allowed to handle this standard.

2.5.1. Building the MPEG Bit Stream

To understand how the component parts of the bit stream are multiplexed, we need to first look at each component part. The most basic component is known as an Elementary Stream in MPEG.

Elementary Stream

Each Elementary Stream (ES) output by an MPEG audio, video and some data encoders contain a single type of signal.

For video and audio, the data is organized into access units, each representing a fundamental unit of encoding. For example, in video, an access unit will usually be a complete encoded video frame.

Packetized Elementary Stream

Each ES is input to an MPEG-2 processor which accumulates the data into a stream of Packetized Elementary Stream (PES) packets. A PES packet may be a fixed or variable sized block, with up to 65536 bytes per block and includes a 6 byte protocol header. A PES is usually organized to contain an integral number of ES access units.

The PES header starts with a 3 byte start code, followed by a one byte stream ID and a 2 byte length field.

MPEG-2 Multiplexing

The MPEG-2 standard allows two ways of multiplexing[16]:

- MPEG Program Stream: A group of tightly coupled PES packets referenced to the same time base. Such streams are suited for transmission in a relatively error-free environment and enable easy software processing of the received data. This form of multiplexing is used for video playback and for some network applications.

- MPEG Transport Stream[17]: Each PES packet is broken into fixed-sized transport packets forming a general purpose way of combining one or more streams, possibly with independent time bases. This is suited for transmission in which there may be potential packet loss or corruption by noise, or / and where there is a need to send more than one program at a time.

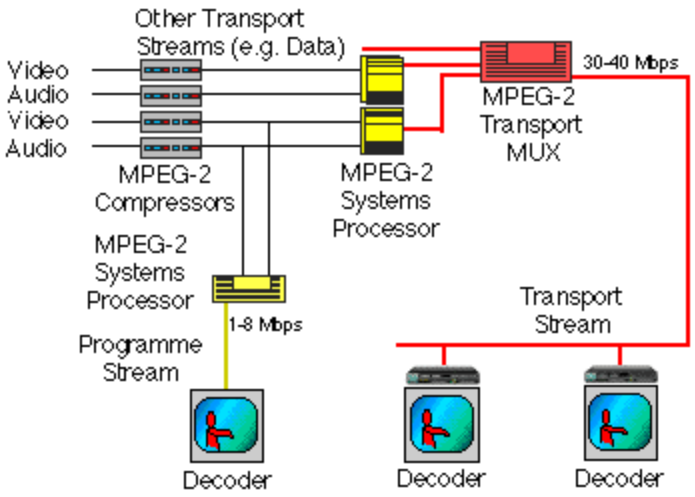


Figure 13. Combining elementary streams from encoders into a TS or a PS

The Program Stream is widely used in digital video storage devices, and also where the video is reliably transmitted over a network (e.g. video-clip download). Digital Video Broadcast (DVB) uses the MPEG-2 Transport Stream over a wide variety of underlying networks. Since both the Program Stream and Transport Stream multiplex a set of PES inputs and the interoperability between the two formats may be achieved at the PES level.

2.5.2. MPEG Transport Stream

A transport stream consists of a sequence of fixed sized transport packet of 188 Bytes. Each packet comprises 184 Bytes of payload and a 4 Bytes header. One of the items in this 4 Byte header is the 13 bit Packet Identifier (PID) which plays a key role in the operation of the Transport Stream.

The format of the transport stream is described using the figure below. This figure shows two elementary streams sent in the same MPEG-2 transport multiplex. Each packet is associated with a PES through the setting of the PID value in the packet header (the values of 64 and 51 in the figure). The audio packets have been assigned PID 64, and the video packets PID 51 (these are arbitrary, but different values).

Figure 14. Single Program Transport Stream (Audio and Video PES)

As is usual, there are more video than audio packets, but we may also note that the two types of packets are not evenly spaced in time. The MPEG-TS is not a time division multiplex, packets with any PID may be inserted into the TS at any time by the TS multiplexor. If no packets are available at the multiplexor, it inserts null packets (denoted by a PID value of 0x1FFF) to retain the specified TS bit rate. The multiplexor also does not synchronize the two PESs. A separate process is therefore required to synchronize the two streams.

Transmission of the MPEG-TS

Although the MPEG TS may be directly used over a wide variety of media, it may also be used over a communication network. It is designed to be robust with short frames, each one being protected by a strong error correction mechanism. It is constructed to match the characteristics of the generic radio or cable channel and expects an uncorrected Bit Error Rate (BER) of better than 10^{10}.

The MPEG-2 Transport Stream is so called, to signify that it is the input to the Transport Layer in the ISO Open System Interconnection (OSI) seven-layer network reference model. It is not, in itself, a transport layer protocol and no mechanism is provided to ensure the reliable delivery of the transported data. MPEG-2 relies on underlying layers for such services and the transport requires the underlying layer to identify the transport packets, and to indicate in the transport packet header, when a transport packet has been erroneously transmitted.

2.5.3. Format of a Transport Stream Packet

Each MPEG-2 TS packet carries 184 B of payload data prefixed by a 4 B (32 bit) header.

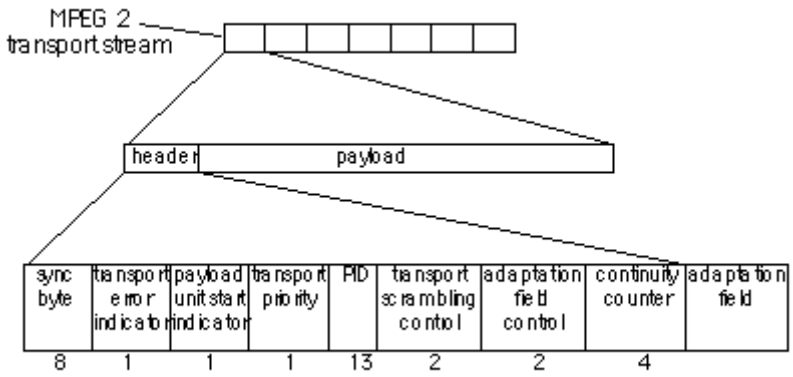


Figure 15. MPEG-2 Transport Stream Header Description

The header has the following fields:

- Synchronization Byte (8 bits). This has the bit pattern 0x47 (0100 0111).

Next, a set of three flag bits are used to indicate how the payload should be processed.

- The first flag indicates a transport error.
- The second flag indicates the start of a payload.
- The third flag indicates transport priority bit.

The flags are followed by a 13 bit Packet Identifier (PID). This is used to uniquely identify the stream to which the packet belongs (e.g. PES packets corresponding to an ES) generated by the multiplexer. The PID allows the receiver to differentiate the stream to which each received packet belongs. Some PID values are predefined and are used to indicate various streams of control information. A packet with an unknown PID, or one with a PID which is not required by the receiver, is silently discarded. The

particular PID value of 0x1FFF is reserved to indicate that the packet is a null packet (and has to be ignored by the receiver).

The two scrambling control bits are used by conditional access procedures to encrypt the payload of some TS packets.

- Two adaption field control bits which may take four values:
    - 01: No adaptation field, payload only.
    - 10: Adaptation field only, no payload.
    - 11: Adaptation field followed by payload.
    - 00: Reserved for future use.

- Finally there is a half byte Continuity Counter (4 bits).

CHAPTER 3. FRAMEWORKS

During this project, there has been an investigation on different frameworks available to form the video wall. The following are the four frameworks to focus on: GStreamer, FFmpeg, VLC and OMXPlayer.

As a brief summary, the following table shows in which system component has been used each framework:

Framework	Server	Client	Function
GStreamer	**Yes**	**Yes**	Stream and receive audio and video flows.
FFmpeg	**Yes**	**No**	Stream video flows.
VLC	**Yes**	**No**	Stream video flows.
OMXPlayer	**No**	**Yes**	Receive audio and video flows.

Table 3. Frameworks and components relationship

As it can be observed in the table above not all the frameworks are available for both, server and client.

In this chapter the most relevant features of each one of these frameworks will be described.

3.1. GStreamer

GStreamer[18] is a library for constructing graphs of media-handling components. The applications it supports range from simple Ogg/Vorbis playback, audio/video streaming to complex audio (mixing) and video (non-linear editing) processing.

Applications can take advantage of advances in codec and filter technology transparently. Developers can add new codecs and filters by writing a simple plugin with a clean, generic interface.

Figure 16. GStreamer framework logo

GStreamer is released under the LGPL (Lesser General Public Licence).

The GStreamer main characteristics are:

Multiplatform

GStreamer has been ported to a wide range of operating systems, processors and compilers. These include but are not limited to Linux on x86, PPC and ARM using GCC. Solaris on x86 and SPARC using both GCC and Forte, MacOS X, Microsoft Windows using MS Visual Developer, IBM OS/400 and Symbian OS.

GStreamer can bridge to other multimedia frameworks in order to reuse existing components (e.g. codecs) and use platform input/output mechanisms:

- Linux/Unix: OpenMAX-IL (via gst-omx)
- Windows: DirectShow
- Mac OS X: QuickTime

Comprehensive Core Library

- Graph-based structure allows arbitrary pipeline construction.
- Based on GLib 2.0 object model for object-oriented design and inheritance.
- Compact core library of less than 500KB, about 65 K lines of code.
- Multi-threaded pipelines are trivial and transparent to construct.
- Clean, simple and stable API for both plugin and application developers.
- Extremely lightweight data passing means very high performance/low latency.
- Complete debugging system for both core and plugin/app developers.
- Clocking to ensure global inter-stream synchronization (a/v sync).
- Quality of service (QoS) to ensure best possible quality under high CPU load.

Intelligent Plugin Architecture

- Dynamically loaded plugins provide elements and media types, demand-loaded via a registry cache.
- Element interface handles all known types of sources, filters and sinks.
- Capabilities system allows verification of element compatibility using MIME types and media-specific properties.
- Auto plugging uses capabilities system to complete complex paths automatically.
- Pipelines can be visualised by dumping them to a .dot file and creating a PNG image from that.
- Resource friendly plugins to do not waste memory.

Broad Coverage of Multimedia Technologies

GStreamer capabilities can be extended through new plugins. The features listed below are just a rough overview of what is available using the GStreamer own plugins, not counting any third party offerings.

- Container formats: asf, avi, 3gp/mp4/mov, flv, mpeg-ps/ts, mkv/webm, mxf, ogg.
- Streaming: http, mms, rtsp, rtp.
- Codecs: FFmpeg, various codec libraries, 3rd party codec packs.
- Metadata: native container formats with a common mapping between them.
- Video: various colorspaces, support for progressive and interlaced video.

- Audio: integer and float audio in various bit depths and multichannel configurations.

Extensive Development Tools

- gst-launch command-line tool for quick prototyping and testing.
- A lot of documentation, including partially completed manual and plugin writer's guide.
- Large selection of test programs and example code in each module.
- Access to GStreamer API with various programming languages.

3.2. FFmpeg

FFmpeg[47] is a free software project that produces libraries and programs for handling multimedia data. FFmpeg includes libavcodec, an audio/video codec library used by several other projects, libavformat, an audio/video container mux and demux library, and the ffmpeg command line program for transcoding multimedia files. FFmpeg is published under the GNU Lesser General Public License 2.1+ or GNU General Public License 2+ (depending on which options are enabled).

Figure 17. FFmpeg logo

FFmpeg is the leading multimedia framework, able to decode, encode, transcode, mux, demux, stream, filter and play a lot of formats. No matter if they were designed by some standards committee, the community or a corporation. It contains libavcodec, libavutil, libavformat, libavfilter, libavdevice, libswscale and libswresample which can be used by applications. As well as ffmpeg, ffserver, ffplay and ffprobe which can be used by end users for transcoding, streaming and playing.

The FFmpeg project tries to provide the best technically possible solution for developers of applications and end users alike. To achieve this the best free software options available are combined. They slightly favor their own code to keep the dependencies on other libs low and to maximize code sharing between parts of FFmpeg. Wherever the question of "best" cannot be answered both options are supported so the end user can choose.

Security is a high priority, and code review is always done with security in mind. Though due to the very large amounts of code touching untrusted data, security issues are unavoidable. And thus they provide as quick as possible updates to the last stable releases when new security issues are found.

Tools

FFmpeg provides various tools:

- *ffmpeg* is a command line tool to convert multimedia files between formats.

- *ffserver* is a multimedia streaming server for live broadcasts.
- *ffplay* is a simple media player based on SDL and the FFmpeg libraries.
- *ffprobe* is a simple multimedia stream analyzer.

Libraries

FFmpeg provides various developers libraries:

- *libavutil* is a library containing functions for simplifying programming, including random number generators, data structures, mathematics routines, core multimedia utilities, and much more.

- *libavcodec* is a library containing decoders and encoders for audio/video codecs.

- *libavformat* is a library containing demuxers and muxers for multimedia container formats.

- *libavdevice* is a library containing input and output devices for grabbing from and rendering to many common multimedia input/output software frameworks, including Video4Linux, Video4Linux2, VfW, and ALSA.

- *libavfilter* is a library containing media filters.

- *libswscale* is a library performing highly optimized image scaling and color space/pixel format conversion operations.

- *libswresample* is a library performing highly optimized audio resampling, rematrixing and sample format conversion operations.

3.3. VLC

VLC media player[54] is a portable, free and open-source, cross-platform media player and streaming media server written by the VideoLAN project. VLC media player supports many audio and video compression methods and file formats, including DVD-Video, video CD and streaming protocols. It is able to stream over computer network and to transcode multimedia files.

The default distribution of VLC includes a large number of free decoding and encoding libraries, avoiding the need for finding/calibrating proprietary plugins. Many of VLC's codecs are provided by the libavcodec library from the FFmpeg project, but it uses mainly its own muxer and demuxers and its own protocols implementations. It also gained distinction as the first player to support playback of encrypted DVDs on Linux and OS X by using the libdvdcss DVD decryption library.

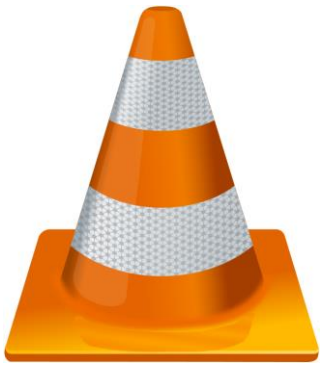


Figure 18. VLC Logotype

Because VLC is a packet-based media player, it plays almost all video content. It can play some, even if they're damaged, incomplete, or unfinished. It also plays MPEG transport streams (.TS) files.

VLC supports all audio and video formats supported by libavcodec and libavformat. This means that VLC can play back H.264 or MPEG-4 Part 2 video as well as support FLV or MXF file formats "out of the box" using FFmpeg's libraries. Alternatively, VLC has modules for codecs that are not based on FFmpeg's libraries.

VLC has some filters that can distort, rotate, split, deinterlace, and mirror videos as well as create display walls or add a logo overlay.

VLC can stream live, unencrypted content to a monitor or HDTV. VLC media player can display the playing video as the desktop wallpaper, like Windows DreamScene, by using DirectX, only available on Windows operating systems.

VLC can be installed or run directly from a USB flash drive or other external drive. VLC can be extended through scripting. It uses the Lua scripting language. VLC can play videos in the AVCHD format, a highly compressed format used in recent HD camcorders. VLC can generate a number of music visualization displays. The program is able to convert media files into various supported formats.

3.4. OMXPlayer

OMXPlayer is a video player specifically made for the Raspberry Pi devices.

It takes advantage of the GPU to decode H.264 videos and it comes with the standard Raspbian Wheeze distribution.

OMXPlayer has been used in the Raspberry Pi devices to play back the streamed videos.

OMXPlayer is installed by default on the Raspberry Pi.

When typing the command **omxplayer**, a list of the available command line options is observed on screen.

```
Usage: omxplayer [OPTIONS] [FILE]
Options :
-h / --help                print this help
-n / --aidx  index         audio stream index: e.g. 1
-o / --adev  device        audio out device: e.g. hdmi/local
-i / --info                dump stream format and exit
-s / --stats               pts and buffer stats
-p / --passthrough         audio passthrough
-d / --deinterlace         deinterlacing
-w / --hw                  hw audio decoding
-3 / --3d mode            switch tv into 3d mode (e.g. SBS/TB)
-y / --hdmiclocksync      adjust display refresh rate to match
                                                video (default)
-z / --nohdmiclocksync do not adjust display refresh rate to match video
-t / --sid index           show subtitle with index
-r / --refresh                 adjust framerate/resolution to
video
-l / --pos                 start position (in seconds)
    --boost-on-downmix     boost volume when downmixing
    --subtitles path      external subtitles in UTF-8 srt form at
    --font path            subtitle font
                           (default:/usr/share/fonts/truetype/
                                        freefont/FreeSans.ttf)
    --font-size size     font size as thousandths of screen height
                                                (default: 55)
    --align left/center    subtitle alignment (default: left)
      --lines n              number of lines to accommodate in
                              the subtitle buffer (default: 3)
```

The following key commands do the following actions when pressed while playing a video in OMXPlayer.

```
z           Show Info
1           Increase Speed
2           Decrease Speed
j           Previous Audio stream
k           Next Audio stream

i           Previous Chapter
o           Next Chapter

n           Previous Subtitle stream
m           Next Subtitle stream
s           Toggle subtitles
d           Subtitle delay -250 ms
f           Subtitle delay +250 ms
q           Exit OMXPlayer
Space or p  Pause/Resume

-           Decrease Volume
+           Increase Volume

Left Arrow  Seek -30
Right Arrow Seek +30
Down Arrow  Seek -600
Up Arrow    Seek +600
```

CHAPTER 4. MILESTONES

This chapter will describe the different milestones defined in this project and how they are achieved by using the different frameworks available.

The work methodology employed is the following:

1. Defining the deployment scenarios.
2. For each scenario, defining different milestones.
3. For each milestone, trying to implement a solution using the available frameworks.
4. Translation of each milestone pipeline to C code.

4.1. Deployment scenario 1

The deployment scenario 1 is the following:

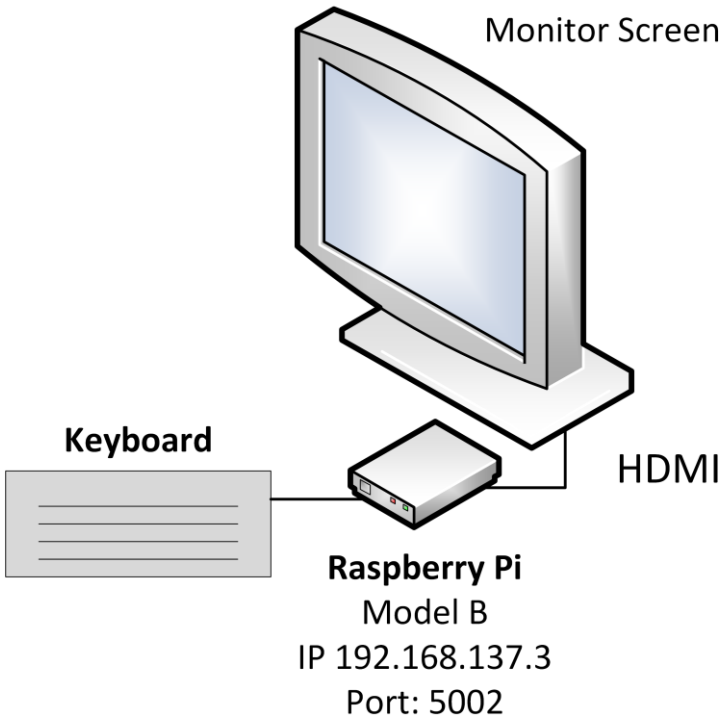


Figure 19. Deployment scenario 1

The scenario is composed by the Raspberry Pi, a keyboard and one monitor screen. In this scenario, the following milestones will be performed:

- Play back a H.264 Full HD Video.

4.1.1. Milestone 1 - Playing back a H.264 Full HD Video

GStreamer

After some investigation and testing, how to create a pipeline able to play back a H.264 1080p video was discovered:

```
gst-launch-1.0 filesrc location= /home/pi/Monsters.mp4 ! qtdemux
   name=dmux dmux.video_0 ! queue ! decodebin ! autovideosink
       dmux.audio_0 ! queue ! aacparse ! faad ! alsasink
```

The description of this pipeline is simple. The command **gst-launch-1.0**[20] allows us to access the Gstreamer command-line tool for quick prototyping and testing interface.

The next element, **filesrc location**, allows the reading of data from a file in the local file system. The path of the file needs to be specified.

The element **!** is used for linking the file to the next element, in this case the **qtdemux** (Quicktime demuxer). **Qtdemux** demultiplex a file into audio and video streams. A name for the demuxer has to be specified. In this case the name is **dmux**. At this point, each part of the stream has to be linked to other elements that will decode and print them on the screen.

To indicate the command of dealing with the video stream **dmux.video_0** is written. This specifies that we will treat now with the video part only. The video stream is linked to a queue by the element **!**. The queue will create a new thread to handle this part of the stream. To decode the video the information stored in the queue will be linked to the element **decodebin** by using !. **decodebin** is an auto-plugin to decode raw-media. It is a container for a collection of elements, in this case, decoders. Essentially, this element will find the correct decoder to decode the video stream. Once the video is decoded the result is linked to the element which will print out the video on the screen, by using !. The element responsible of it is **autovideosink.** It detects an appropriate video sink to use.

Having coped with the video stream, the audio stream remains to be handled. To deal with the audio stream **dmux.audio_0** has to be added next to **autovideosink** without ! since **autovideosink** is not to be linked to another element. This indicates that we want to treat the audio stream. The audio stream is linked to a new queue that will create a thread to handle in parallel the audio part. The next step is to link this to the element **aacparse** which is a parser for Advanced Audio Coding that handles both ADIF and ADTS stream formats. The audio is now ready to be decoded. GStreamer has an AAC decoder element called Free – Advanced Audio Decoder (**faad**) in charge of decoding aac audio format. Finally the result of the decoder needs to be linked to the element **alsasink**. This element provides an output to a sound card via ALSA.

FFmpeg

Due to there is no HW Support of FFmpeg in the Raspberry Pi, it was decided to use the GStreamer and OMXPlayer reception to play back the videos on the Raspberry Pi. The same reason applies to the rest of reception milestones in this chapter.

VLC

There is HW Support of VLC in the Raspberry Pi[60] but due to the complexity and the constant request of multiple third party libraries, this option was impossible to implement. 2 days (7 hours per day) were devoted to it but the compilation failed. All the installation details are explained in Annex C. The final decision was to use the GStreamer and OMXPlayer reception to play back the videos on the Raspberry Pi.

OMXPlayer

The following pipeline plays back nicely a Full HD video stored in the Raspberry Pi device. The audio is outputted through the speakers connected to the Raspberry Pi device.

```
omxplayer Monsters.mp4
```

4.2. Deployment scenario 2

The deployment scenario 2 is the following:

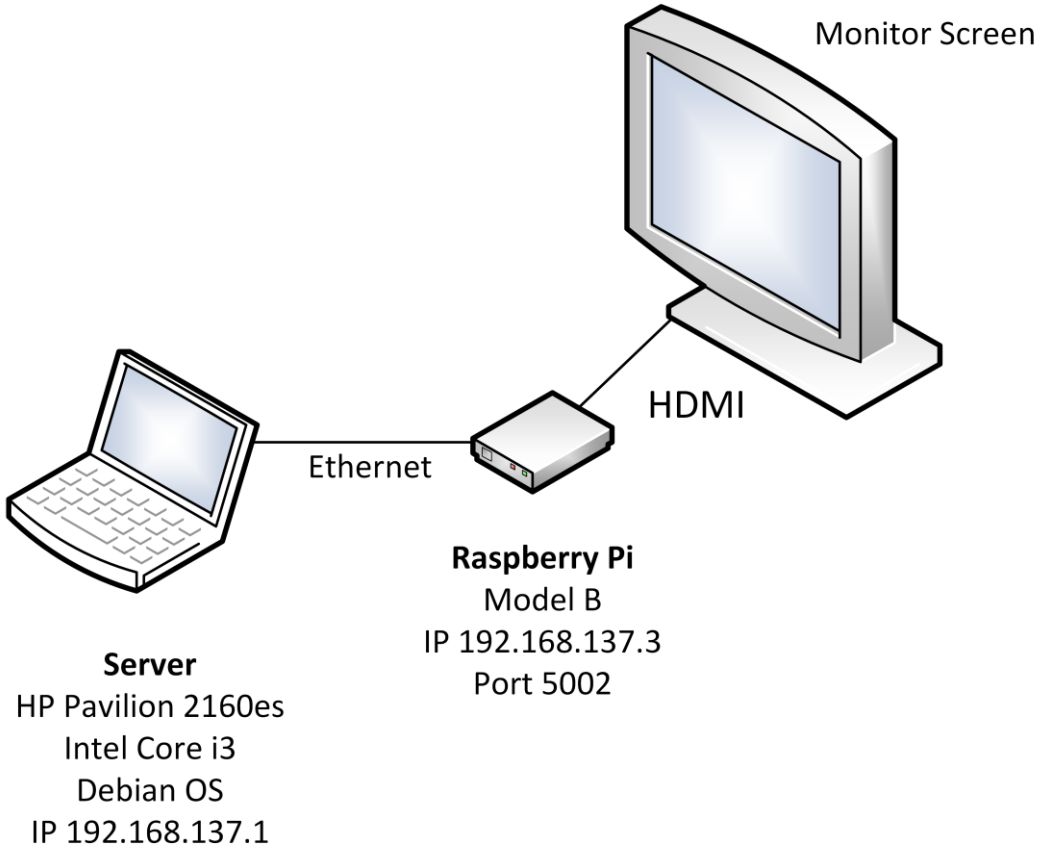


Figure 20. Deployment scenario 2

The scenario is composed by a computer that will act as a server, an Ethernet Cat 5e cable to connect the computer with the Raspberry Pi, a HDMI cable and a monitor screen.

In this scenario, the following milestones will be performed:

- Sending a HD/Full HD video to one Raspberry Pi.
- Sending a real time web cam video flow to one Raspberry Pi.
- Receiving a Full HD video / WebCam and playing back.
- Sending a Full HD video using MPEG-2 TS.
- Receiving a Full HD video using MPEG2-TS.

4.2.1. Milestone 2 - Sending a HD/Full HD video to one Raspberry Pi

GStreamer

The aim is to find out how to stream a H.264 Full HD video over the network using the RTP protocol.

```
gst-launch-1.0 filesrc location=Videos/Monsters.mp4 ! qtdemux
name=dmux dmux.video_0 ! queue ! rtph264pay config-interval=1 !
udpsink host=192.168.137.3 port=5002  dmux.audio_0 ! queue !
aacparse ! faad ! alsasink
```

The aim is to describe only the new elements that appear above since most of them are described in Milestone 1.

The procedure is to separate the video stream and the audio stream and then encapsulate them into RTP packet. After, stream them using UDP to the receiver. The audio is playing in local because the **faad** element belongs to bad-plugins and its behaviour when receiving the audio stream through the network was unpredictable. For this reason it was decided to play back the audio in local, to avoid troubles.

As it can be observed, the pipeline starts as in Milestone 1. Once the video stream is demultiplexed it is encapsulated into RTP container.

rtph264pay needs to be configurated with a config-interval=1 to send the SPS (Sequence Parameter Set) and PPS (Picture Parameter Set) every 1 second so that the application can decode the content that is coming in. It is a very important parameter. Without indicating a config-interval, the receiver is not able to decode the stream. Chapter 5 explains SPS and PPS in detail.

Finally, the element **udpsink** is the UDP packet sender that sends data over our network via UDP. The destination IP address and the port to send the packets to have to be indicated.

FFmpeg

1. HD video

The aim is to find out how to stream a H.264 HD video over the network using the RTP protocol.

```
ffmpeg -re -i /home/carlos/Descargas/HD/Serenity.mp4 -vcodec
   libx264 -tune zerolatency -preset superfast -an -f rtp
               rtp://192.168.137.3:5002
```

To begin writing a pipeline the command **ffmpeg** has to be used.

The parameter **–re** allows ffmpeg to read input at native frame rate. It is mainly used to simulate a grab device or live input stream (e.g. when reading from a file). Should not be used with actual grab devices or live input streams (where it can cause packet loss). By default ffmpeg attempts to read the input(s) as fast as possible. This option will slow down the reading of the input(s) to the native frame rate of the input(s). It is useful for real-time output (e.g. live streaming).

The parameter **–i** is used to specify the input format file or device. The path of the file that is to be streamed is indicated.

The parameter **–vcodec** will set the video codec used for encoding the video file. In this case **libx264**[50] is used. It is a free software library and application for encoding video streams into the H.264/MPEG-4 AVC compression format, and is released under the terms of the GNU GPL. Then two parameters are added (speed-preset and tune, which are exactly the same as GStreamer). The process of decoding and encoding is done when the library is called.

The step of decoding and re-encoding the video could be omitted with the option **–vcodec copy** which only will demux and mux. This could be very useful to reduce CPU usage in the server as happens with GStreamer.

Following this way, the output stream does not work with the reception pipeline and nothing is shown on screen. After a lot of testing trying to obtain the same result as GStreamer, all the attempts are unsuccessful. The problem remains in the transmission of SPS and PPS messages. In FFmpeg captures with Wireshark messages are not seen. Once the video is re-encoded, the SPS and PPS messages appear in the capture.

Next to it, the parameter **–an** is indicated to skip audio.

Finally, the indication of the output file (**-f**) is required. In this case, the output of the encoded stream must be encapsulated over RTP[52] and sent to the IP and port indicated in the pipeline.

When the pipeline is executed the following informations appears on the screen:

```
Press ctrl-c to stop encoding
frame=  326 fps= 24 q=23.0 size=    3489kB time=13.60 bitrate=2102.2kbits/s
```

Figure 21. FFmpeg statistics Milestone 1

Here some interesting information is observed. First of all, FFmpeg is saying which frame is now streaming. Secondly, it provides information about the frame rate. In this example, the video has been previously analyzed and it is known that its frame rate is 24 fps. Hence, FFMpeg is streaming the video with a correct frame rate.

Then FFmpeg provides us the parameter q=23. This is related to the Constant Rate Factor (CRF). CRF allows the encoder to achieve a certain output quality for the whole file when output file size is of less importance. This provides maximum compression efficiency with a single pass. Each frame gets the bitrate it needs to keep the requested quality level. The downside is that it is not possible to tell FFmpeg to get a specific file size or not to go over a specific size or bitrate.

The range of the quantizer scale is 0-51: where 0 is lossless, 23 is default, and 51 is worst possible. A lower value means a higher quality and a subjectively sane range is 18-28. Considered 18 to be visually lossless or nearly, it should look the same or nearly the same as the input but it is not technically lossless.

The range is exponential, so increasing the CRF value +6 is roughly half the bitrate while -6 is roughly twice the bitrate. General usage is to choose the highest CRF value that still provides an acceptable quality. If the output looks good, then try a higher value and if it looks bad then choose a lower value.

Therefore, here q=23 means default value of the quantizer and the quality of the video in reception is good.

Finally, the size of the video streaming sent to the network (in KBytes), the elapsed time and the bit rate are shown.

2. Full HD video

The aim is to find out how to stream a H.264 Full HD video over the network using the RTP protocol. It is the next step of the previous point. Since point 1 has worked correctly, the same is going to be done but using a Full HD video.

The pipeline is the following:

```
ffmpeg -re -i /home/carlos/Descargas/FullHD/Monsters.mp4 -vcodec
     libx264 -tune zerolatency -preset superfast -an -f rtp
                    rtp://192.168.137.3:5002
```

Note that it is exactly the same pipeline, being the video the only change (instead of a 720p, a 1080p video is used).

However, the results obtained are not the same as in the previous case (720p video):

```
Press ctrl-c to stop encoding
frame=   90 fps= 13 q=23.0 size=    3451kB time=3.75 bitrate=7532.0kbits/s
```

Figure 22. FFmpeg statistics Full HD video

There are differences in some of the parameters comparing them to the previous case. First, the bit rate increases much compared to the previous case (HD), which is normal because it is Full HD video, whose avarage bit rate is usually between 8 and 12 Mbps. On the contrary, the bit rates of the HD videos fluctuate between 3 and 5 Mbps.

The maximum peek is not reflected on the two different pictures above, though peeks about 11 Mbps, in the case of Full HD videos, and 5 Mbps, in case of HD videos, were seen.

The most interesting parameter is that FFmpeg is not able to maintain the original frame-rate of the video (24 fps) and it maintains it around 13 fps, with some fluctuations. Although, it has never arrived to 24 fps in any of the tests done.

Different solutions has been tried. The first has been setting the CRF value manually. Using the parameter –crf in the encoder, it is possible to decrease or increase the

value. The value increases up to 28 (the limit value inside the sane range) and the results are the following:

```
Press ctrl-c to stop encoding
frame=  468 fps= 21 q=28.0 size=    1910kB time=19.52 bitrate= 801.6kbits/s
```

Figure 23. FFmpeg statistics (CRF applied)

As expected, bitrate decreases and the fps increases but it is still not enough to fluently play back the video in reception. Increasing CRF up to insane value, correct fps rate is achieved but quality becomes a disaster.

The other 'solution' is to manually fix the bit rate to certain amount using the parameter **–b** and the maximum bitrate value. After trial and error, it is found that around 2500 Kbit/s bit rate the fps value increases up to 24. However, again, this reduces quality and the visualization is not as good as with the GStreamer emmiter.

VLC

The following command takes the video and sends it directly to the receiver. No decoding or re-encoding is needed.

```
cvlc Serenity.mp4 --sout '#rtp{ dst=192.168.137.3,port-video=5002}'
```

The command is easy to understand: cvlc (command vlc) is the command that allows to create the pipelines. The first parameter is the video file to stream. Then –sout (stream output) indicates the type of output (in our case RTP protocol). Furthermore the destination address and the video port are indicated. The receiver is listening to the same IP and port to play back the stream.

The problem is that the receivers are not able to decode perfectly the stream. The stream is printed in the screen but errors during the visualization occur. The reception is done using the GStreamer pipeline defined in Milestone 4 in this chapter.

Investigating some Wireshark captures were taken and it was discovered that PPS/SPS were not sent periodically and even more, no trace of I, P and B frames appeared in the capture. Instead of them, some mark undefined packets appeared, as in figure 24. Deeper analysis of Wireshark captures is described in chapter 5.

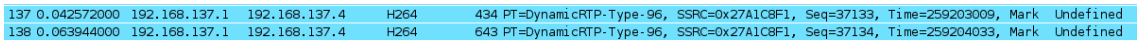


Figure 24. Mark undefined packets – VLC

An effort was made in order to send a Full HD video using the same pipeline but quality playback in reception was very low.

4.2.2. Milestone 3 - Sending a real time web cam video flow to one Raspberry Pi

This milestone has been implemented only with GStreamer framework.

The aim is to find out how to stream a WebCam video flow from the server to the client. The resolution of the server web cam is 640x480.

```
gst-launch-1.0 v4l2src ! 'video/x-raw, format=YUY2, width=640,
height=480' ! x264enc tune="zerolatency" speed-preset=2 ! queue
      ! rtph264pay ! udpsink host=192.168.137.3 port=5002
```

The goal is to describe only the new elements since most of them are described in Milestone 1 and 2.

A new element that appears here is **v4l2src.** This element reads frames from a Video4Linux2 device. A web cam is a type of v4l2src so this element sets up the webcam and reads frames from it. The features of the web-cam have to be linked to the element. In this case, the web-cam generates unstructured and uncompressed raw video data using format YUY2 Video picture encoding, with a resolution of 640x480.

The format of the Web Cam video is not H.264 so encode needs to be done using the element x264enc.

x264enc is the H.264 encoder. Some parameters have been set to it. One of them is **speed-preset**. That is the preset name for speed/quality trade-off options (taking into account that can affect decode compatibility). In this case this option is set to the value 2, which means super-fast. In case of having a better server this parameter can be switched to value 3-4. If value 3 is used in the server the CPU usage increases up to 90% and saturates so, speed-preset with value 2 becomes the better value to use.

The second parameter of x264enc is **tune**. Its value is **zerolatency**. Reducing x264's latency is possible, but reduces quality. Since latency in the encoder is not wanted, this parameter is set up.

4.2.3. Milestone 4 - Receiving a Full HD video / WebCam and playing back

This milestone has been implemented only with GStreamer framework.

This pipeline describes the inverse process done in Milestones 2 and 3. The aim is to decode the H.264 encoded video in the server and play it back in the receiver without latency and without reducing quality.

```
gst-launch-1.0 udpsrc uri=udp://192.168.137.3:5002
caps="application/x-rtp, clock-rate=90000, payload=96" !
rtpjitterbuffer latency=500 ! rtph264depay ! decodebin !
autovideosink
```

The aim is to describe only the new elements that appear here since most of them are described in Milestone 1, 2 and 3.

In this pipeline **udpsrc**[29] appears as a first element. It receives specified data over the network via UDP from the IP and port. A parameter called **caps** is added to udpsrc. The first caps parameter indicates that the type of content is RTP, then clock-rate and payload type refers to RTP audio video profile (RTP/AVP). A clock-rate of 90000 and payload type equal to 96 refer to H.264 default payload type.

The next element is **rtpjitterbuffer** that is responsible of dealing with network jitter and other transmission faults. The rtpjitterbuffer will wait for missing packets up to a configurable time limit using the latency property.The default latency is set to **150 ms**. For Full HD videos the value needs to be increased up to **500 ms**.

Then the element **rtph264depay** (RTP H.264 depayloader) will be used to extract H.264 video from RTP packets.

Finally, it is important to comment that the **decodebin** element will find the appropriate decoder for the H.264 video. It will find the decoder **omxh264dec**, the GStreamer plugin that allows H.264 hardware decoding using Raspberry Pi.

4.2.4. Milestone 5 - Sending a Full HD video using MPEG-2 TS

GStreamer

The aim of this pipeline is to be able to send the video using a RTP MPEG-2 TS container. When done, it will be possible to use this project for other types of receivers that use this type of container.

```
gst-launch-1.0 filesrc location= Descargas/FullHD/Monsters.mp4 !
 decodebin ! x264enc speed-preset=2 tune=zerolatency ! video/x-
   h264 ! mpegtsmux ! rtpmp2tpay ! udpsink host=192.168.137.3
                            port=5002
```

The main point is to describe only the new elements that appear here since most of them are described in the previous milestones.

The idea is the same as the one shown in Milestone 2. In this case a RTP payloader that does not allow setting a config-interval is used to send periodically SPS and PPS frames. To avoid the problem, decode and re-encode must be done. It is a nonsensical step, though it is mandatory in order to decode the video in our Raspberry Pi devices.

After defining the location of the video, it is decoded using **decodebin** element and re-encoded again using the **x264enc** element. This will increase the CPU usage much because decode and encode do need it.

Another new element is **mpegtsmux**. It multiplexes media streams (in this case, it will mux video and audio streams) into an MPEG Transport Stream. The type of video must be indicated (in this case, **video/x-h264**). This element adds latency to the stream. Finally, as commented above, **rtpmp2tpay** encapsulates payload MPEG2 TS into RTP packets.

VLC

The same idea as in GStreamer case is followed now. Using the GStreamer pipeline shown in Milestone 6, the results obtained were very low quality playback in reception (errors appeared in the video playback).

```
cvlc Serenity.mp4 --sout '#rtp{dst=192.168.137.3, port=5002, mux=ts}'
```

After some research, it was found that the problem might be due to SPS/PPS. SPS/PPS had to be related to the use of SDP files, in which transmission parameters could be defined.

A SDP (Session Description protocol) file was created and attached to the command. Here is the command line used to include the SDP file:

```
  cvlc Serenity.mp4 --sout '#rtp{dst=192.168.137.3, port=5002,
          mux=ts, sdp=file://home/carlos/File.sdp}'
```

The results obtained with VLC were not satisfactory, even using the SDP file.

4.2.5. Milestone 6 - Receiving a Full HD video using MPEG2-TS

GStreamer

This pipeline describes the inverse process done in Milestone 5. The aim is to decode the H.264 video over RTP MPEG2-TS container encoded in the server and play it back in the receiver without reducing quality.

```
gst-launch-1.0 udpsrc uri=udp://192.168.137.3:5002
caps="application/x-rtp, clock-rate=90000, payload=96" !
rtpjitterbuffer latency=500 ! rtpmp2tdepay ! decodebin !
autovideosink
```

The new element here is the **rtpmp2tdepay** that extracts MPEG2-TS from RTP packets.

OMXPlayer

OMX player is only able to play back video streams that use MPEG2-TS container. To receive the stream the following pipeline has to be typed:

```
omxplayer udp://192.168.137.3:5002
```

The emmiter pipeline used is the one described in Milestone 5 (GStreamer).

The video stream was not played back correctly. Video errors appeared on the screen. The log files and Wireshark captures of each of the milestones were printed in order to find out the differences. After analyzing them, it was discovered that the local video had no B frames while the streamed video did.

It was then decided to remove B frames from the streamed video. However, the results were not satisfactory. A deeper investigation should be done in order to find out the correct way to stream the video with GStreamer to be received correctly by OMXPlayer.

4.3. Deployment scenario 3

The deployment scenario 3 is the following:

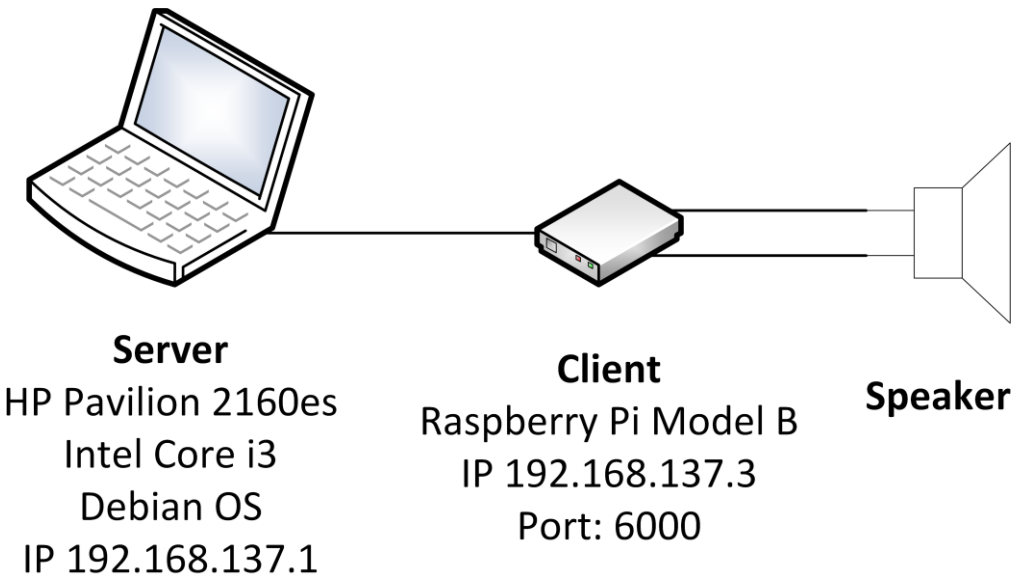


Figure 25. Deployment scenario 3

This is the first deployment scenario in which the audio will be streamed and processed. In the figure above the speaker is connected to the Raspberry Pi device using a 3.5 mm jack to provide the stereo audio.

The following milestones will be performed in this deployment:

- Extracting and sending the audio stream from a 1080p video
- Receiving and playing back the Full HD audio stream

4.3.1. Milestone 7 - Extracting and sending the audio stream from a Full HD video

This milestone has been implemented only with GStreamer framework.

As commented in Milestone 2, some problems with audio stream appeared. After investigating, it was found how to stream the audio contained in a Full HD video correctly. The pipeline able to do it is the following:

```
gst-launch-1.0 filesrc location=Videos/DesMe2.mp4 ! qtdemux
name=dmux dmux.audio_0 ! queue ! decodebin ! audioconvert !
audio/x-raw ! rtpL16pay ! udpsink host=192.168.137.3 port=6000
```

This pipeline uses **qtdmux** for demultiplex the video file into audio. The audio stream will be handled and decoded. Once it is decoded, the aac audio will be converted to 'raw audio' using the element **audioconvert .** This element converts audio into different formats. Finally, RTP L16 Payloader encapsulates raw audio into RTP packets according to RFC 3551 (RTP Profile for Audio and Video Conferences with Minimal Control).

4.3.2. Milestone 8 - Receiving and playing back the Full HD audio stream

This milestone has been implemented only with GStreamer framework.

The inverse process of Milestone 7 is achieved using the following pipeline:

```
gst-launch-1.0 udpsrc uri=udp://192.168.137.3:6000 !
"application/x-rtp,media=(string)audio, clock-rate=(int)96000,
encoding-name=(string)L16, payload=(int)96" ! rtpjitterbuffer !
rtpL16depay ! audioconvert ! alsasink
```

To receive the audio the element **udpsrc** is used and a set of caps parameters need to be used indicating that what it will received is a RTP application which contains audio media, with a clock-rate of 96000, encoded using L16 container and with payload value 96. Then the **rtpjitterbuffer** is used and we decapsulate the audio stream using the element **rtpL16depay**. We convert the audio stream to the one suitable for the audio output (alsasink).

4.4. Deployment scenario 4

The deployment scenario 4 is the following:

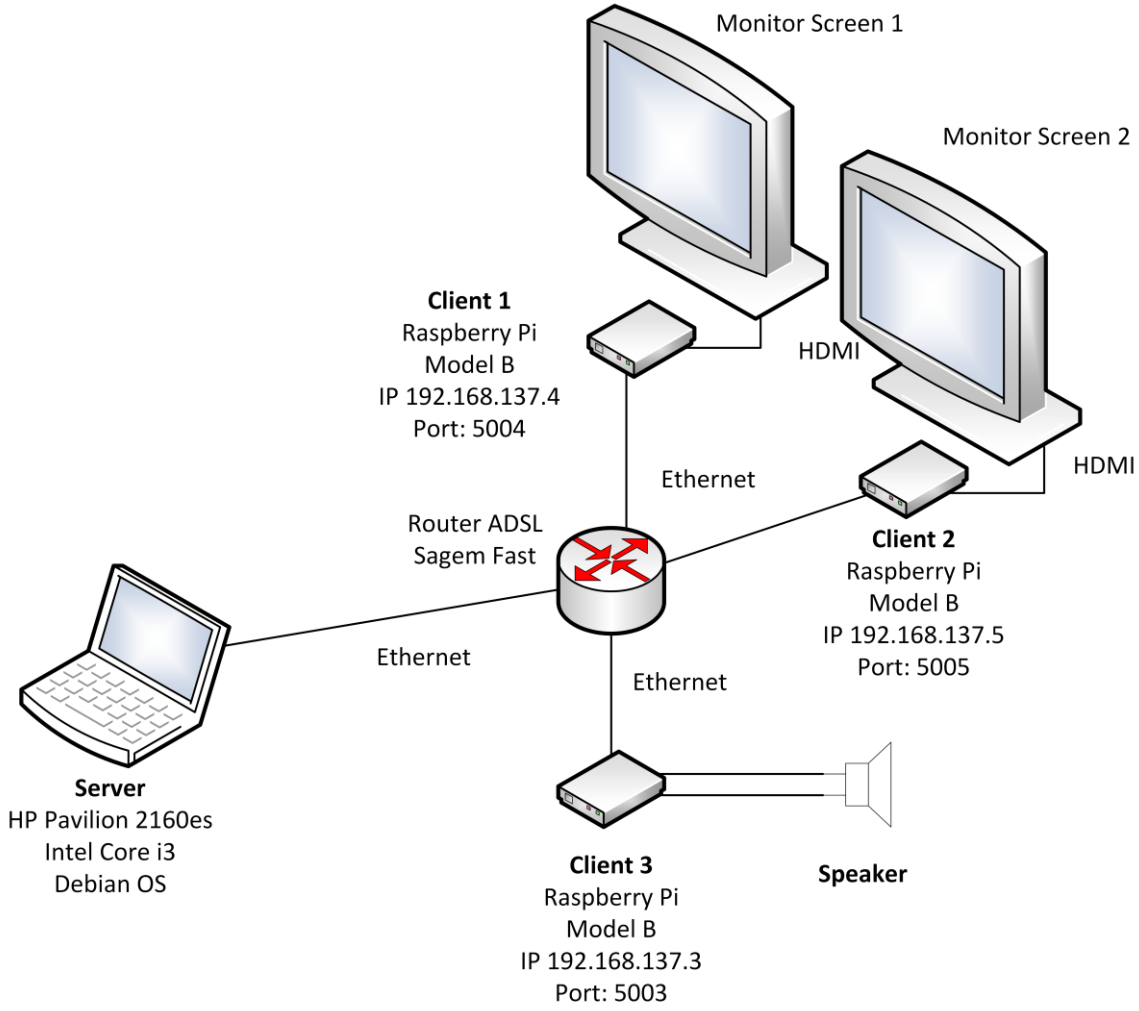


Figure 26. Deployment Scenario 4

This is the last deployment scenario that combines all the previous ones. It is composed by the server, a router to forward the audio and video flows to the corresponding Raspberry Pi device and monitor screens and speakers to visualize the video and listen to the audio.

In this scenario, the following milestones will be performed:

- Splitting and sending Full HD video to Raspberry Pi devices
- Receiving a Full HD/Cam and playing back / cropping the video

4.4.1. Milestone 9 - Splitting and sending Full HD video to Raspberry Pi devices

GStreamer

In this case there are 2 Raspberry Pi, each one connected to a different monitor screen. In addition a router (or switch) is needed in between to send two video flows (one to each Raspberry). It is necessary because our server only has one Ethernet port.

The aim of the pipeline defined below is to split the video into two different video flows and send them to the respectively Raspberry Pi devices. The two Raspberry Pi devices will use the pipeline defined in Milestone 4.

```
gst-launch-1.0 filesrc location=Descargas/FullHD/WalkingDead.mp4
   ! qtdemux name=dmux dmux.video_0 ! queue ! decodebin ! tee
   name=t ! queue ! videocrop right=960  bottom=312 ! x264enc
speed-preset=2 tune=zerolatency ! rtph264pay config-interval=1 !
  udpsink host=192.168.137.4 port=5004 t. ! queue ! videocrop
left=960 bottom=312 ! x264enc speed-preset=2 tune=zerolatency !
   rtph264pay config-interval=1 ! udpsink host=192.168.137.5
  port=5005 dmux.audio_0 ! queue ! aacparse ! faad ! alsasink
```

The pipeline starts, as usual, defining the video that will be streamed to the network. The element **qtdemux** demultiplex the video file into audio and video streams. The video stream will be handled and decoded. It is mandatory to decode in oder to apply video filters.

Since the video flow has to be duplicated to be sent to two different receivers, the element **tee** with name **t** is needed. This element splits data to multiple branches. Branching the data flow is useful when capturing a video shown on the screen and also encoded and written to a file or need to be streamed through the network.

The next step is to crop the video. A queue generates a thread that will handle the split part of the stream. The next element, **videocrop** (a video filter), can remove parts of the picture on the left, right, top or bottom and output a smaller picture than the input one without the unwanted parts of the border.

In the case of having a 1920x1080p video, four videos with a resolution of 960x540 are needed to maintain the original aspect ratio. This will allow visualizing each part of the video in Full Screen in the corresponding monitor. Due to there are only 2 Raspberry Pi devices available, the video wall can only be performed with 2 monitor screens.

Scalability is possible. For instance, if there were 9 Raspberries it should be possible to implement a video wall of 3x3. The only need is to crop the corresponding part of the

original video for each Raspberry. Obviously, the server must have better features than the ones used in this project.

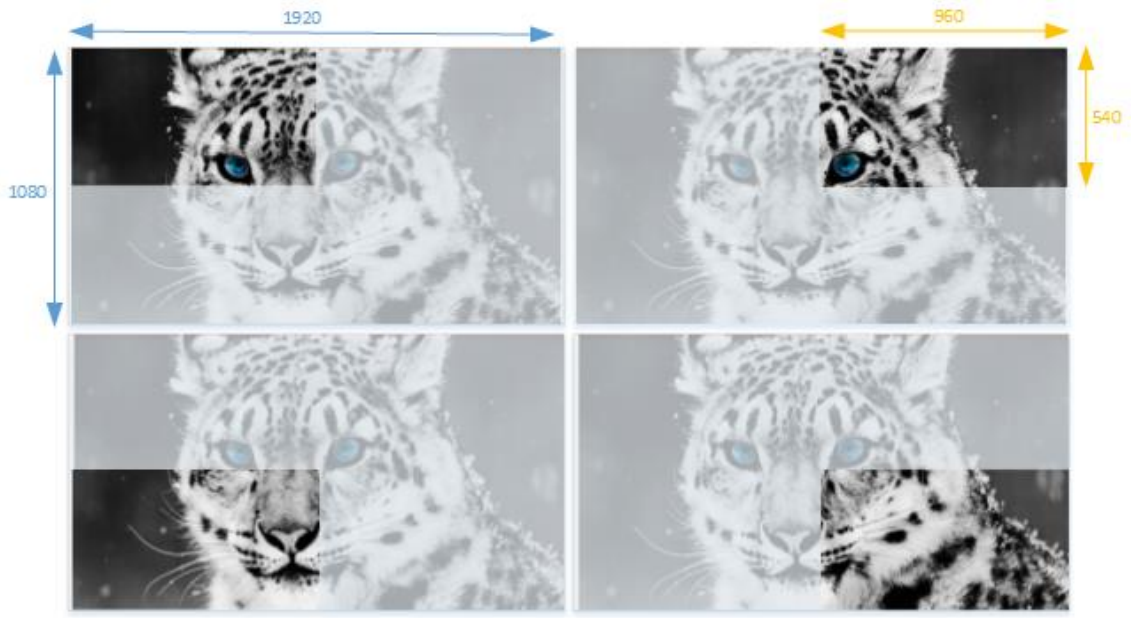


Figure 27. Video cropping example

Now decoding is done using the H.264 encoder and the first video stream is encapsulated and sent through the network.

Exactly the same is done in parallel for the other part of the video. The element **t** is used to separate the two video flows contained in the same pipeline.

CPU usage will increase due to decoding and re-encoding process. It is mandatory to decode and re-encode because when applying a video filter such as a videocrop, the flow must be decoded. Then, once the video filter is applied, re-encode must be done again so that the Raspberry Pi device decodes the stream and shows it on the screen. The difference in CPU usage between Milestone 2 and Milestone 7 is huge but it is due to this intermediate step.

Two alternative ways are possible in order to avoid high CPU usage.

The first one is to pre-crop all the parts of the video using GStreamer. In this case it would be very easy to send all the different flows and the CPU usage would be very low compared with the actual case. The issue of this option is that cropping a video might take a lot of time depending on its length.

The second way is to crop the videos in the Raspberry Pi devices. This method will increase the CPU usage in the receivers. How to crop in the receivers is explained in Milestone 10.

FFmpeg

The aim of the pipeline defined below is to split the video into two different video flows and send them to the respective Raspberry Pi devices.

It is important to say that when using a Full HD video, instead of HD video, the same problem that in Milestone 2 happened. The fps ratio was not enough so this pipeline had to be used with 720p videos.

```
ffmpeg -re -i /home/carlos/Descargas/HD/Serenity.mp4 -vcodec
  libx264  -tune zerolatency -preset superfast -map 0:0 -vf
"crop=640:360:0:0" -an -f rtp rtp://192.168.137.4:5004 -vcodec
  libx264  -tune zerolatency -preset superfast -map 0:0 -vf
   "crop=640:360:640:0" -an -f rtp rtp://192.168.137.5:5005
```

This pipeline is similar to the ones explained in milestone 2. The focus will be on the new elements.

The element -**map** is mapping the input stream to the output stream.

The element **–vf** (video filter) creates a filter graph specified by the input parameter (in our case “crop=640:360:0:0”) and uses it to filter the stream. The parameter indicates that a section of 640x360 pixels is to be cropped in position x=0, y=0. In the second part of the stream the same will be done in position x=640, y=0. When applying these two crops, the first and the second quarters (half of the video) are obtained and sent to the corresponding Raspberry Pi device.

The rest of the elements have been explained in Milestone 2.

It is observed that rame rate is maintained constant to 24 fps and bit rate decreases with respect Milestone 2.

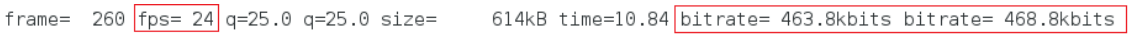


Figure 28. FFmpeg statistics

4.4.2. Milestone 10 - Receiving a Full HD/Cam and playing back / cropping the video

This milestone has been implemented only with GStreamer framework.

The pipeline is very similar with to one described in Milestone 4.

```
gst-launch-1.0 udpsrc uri=udp://192.168.137.4:5004
caps="application/x-rtp, clock-rate=90000, payload=96" !
rtpjitterbuffer latency=500 ! rtph264depay ! decodebin ! queue !
videocrop left=960 bottom=540 ! autovideosink
```

After decoding the video, it is possible to apply videofilters and, in this case, the element videocrop was applied. The result is outputted to the autovideosink element that will display the cropped video.

As commented before, in Milestone 9, this new step implies higher CPU usage in the Raspberry Pi. In chapter 6, the CPU usage obtained results are commented.

Raspberry Pi can only handle Full HD video flows. Therefore, if higher resolution videos (2K, 4K) were to be cropped, this method would not work because Raspberry Pi would not be able to decode them. In the case of 2K or 4K videos, the best choice is to use the Pipeline defined in Milestone 9 using a server with a better CPU than the one used in this project.

4.4.3. Milestone 11 - Combining Milestone 2 and Milestone 7

This milestone has been implemented only with GStreamer framework.

The idea is to be able to send video and audio stream together and synchronized. In order to do this, we need to combine Milestone 2 and Milestone 7. The resulting pipeline is the following:

```
gst-launch-1.0 filesrc location=Videos/Monsters.mp4 ! qtdemux
name=dmux dmux.video_0 ! queue ! rtph264pay config-interval=1 !
udpsink host=192.168.137.255 port=50000 dmux.audio_0 ! queue !
decodebin ! audioconvert ! audio/x-raw ! rtpL16pay ! udpsink
host=192.168.137.3 port=5003
```

Here is the final pipeline obtained that sends the H.264 Full HD video flow in broadcast to all the Raspberry Pi devices in the scenario. The Raspberry Pi devices will decode the stream using the pipeline defined in Milestone 10. Each one of them will have a different value to crop the video.

An extra Raspberry Pi will be needed to play back the audio stream. In the above pipeline the user will indicate the IP address of the Raspberry Pi device that will handle audio stream. When testing a Raspberry Pi device to play back Full HD video and audio at the same time and it worked with some difficulties.

Therefore, to avoid problems, it is recommended to use one extra Raspberry to handle the audio stream perfectly. The pipeline for handling the audio stream in the Raspberry is defined in Milestone 8. It is necessary to remark that the CPU usage of this pipeline

is similar to the one defined in Milestone 2. For this reason it is not taken into account in the CPU comparison in chapter 6.

4.5. Translation of GStreamer Pipelines to C Code

GStreamer pipelines created in different milestones have been translated to C Code. The advantages of writting the code in C are that the programs run quickly because they can access low level facilities (both, in the Raspberry and in the server).

To compile the programs created, the following command is needed:

```
gcc -Wall program.c -o program $(pkg-config --cflags --libs gstreamer-1.0)
```

In the example above a program called *program.c* is being compiled. GStreamer makes use of **pkg-config** to get compiler and linker flags needed to compile this application.

We can run the application by using **./program.**

CHAPTER 5. SYNCHRONIZATION

In a Video Wall the idea of synchronizing a set of screens is that you can use individual content on each screen but still get the monitors to work together in a synchronized way. In the figure below there are three animations (video clips) that are synchronized to start simultaneously and maintain this synchronization during all the streaming.

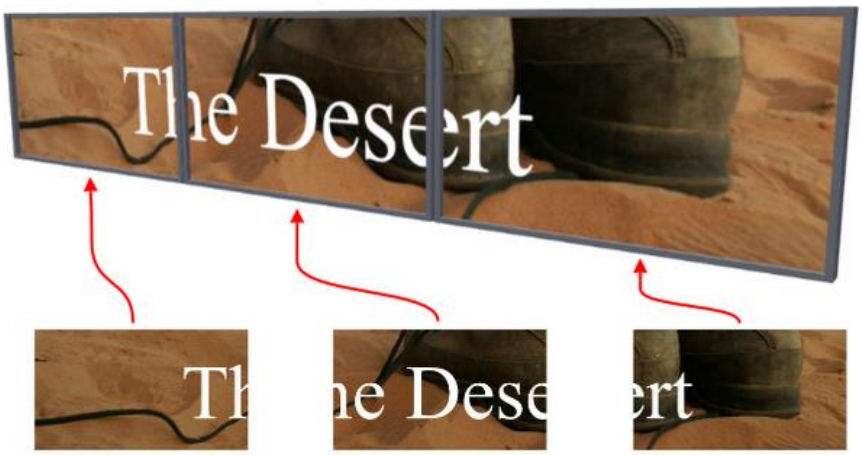


Figure 29: Video Wall syncronization example

The purpose of this chapter is to verify the synchronization between the two Raspberry Pi devices that are used during the project. To do so, Wireshark, a network analyzer tool, will be used to explain the basis of the packets captured. The synchronization evidences shown in this chapter are achieved with GStreamer and FFmpeg frameworks.

5.1. GStreamer

In this case synchonization[30] in the Video Wall will be demonstrated by using GStreamer.

First of all, the GStreamer pipeline statistics indicate that there is just one clock[42] handling the 2 video flows and another clock that handles the local audio.

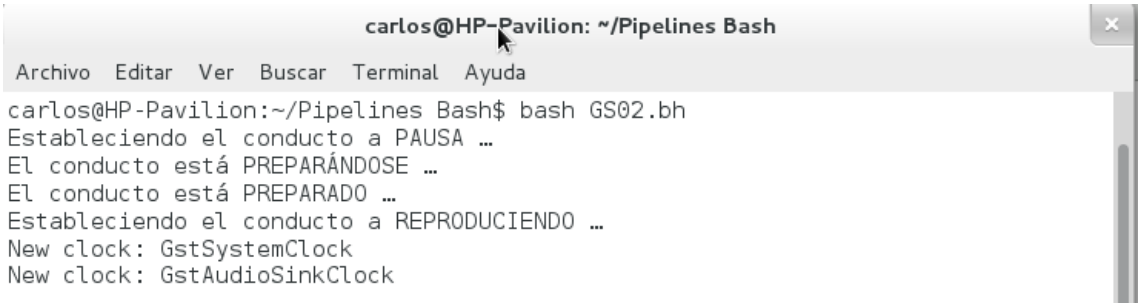

carlos@HP-Pavilion: ~/Pipelines Bash
Archivo Editar Ver Buscar Terminal Ayuda
carlos@HP-Pavilion:~/Pipelines Bash$ bash GS02.bh
Estableciendo el conducto a PAUSA …
El conducto está PREPARÁNDOSE …
El conducto está PREPARADO …
Estableciendo el conducto a REPRODUCIENDO …
New clock: GstSystemClock
New clock: GstAudioSinkClock

Figure 30. GStreamer System clock and audio clock

Second of all, when observing the Wireshark capture below, the Synchronization Source parameter (SSRC) is the same for both flows.

Third of all, focusing on the capture, there are some packets called NAL unit (Network Abstraction Layer). As explained in chapter 2, the encoded video data is organized in NAL units and each NAL contains a header that indicates the type of data and the payload of the data defined in the header.

In the wireshark captures showed below different types of NAL units are seen. In chapter 4, we have seen that it was very important to set a config interval in the element **rtph264pay** to periodically send SPS (Sequence Parameter Set) and PPS (Picture Parameter Set) packets.

SPS and PPS packets (a type of NAL unit) contain information such as picture size, optional coding modes employed and macro block to slice group map. For the importance of information contained, these packets are fundamental to decode correctly the streams in the clients. SPS and PPS packets are sent every 1 second.

Another type of NAL unit, called Supplemental enhancement information packets (SEI) contains timing information and other supplemental data that may enhance usability of the decoded video signal.

Moving on, IDR[9] frames and non IDR frames are observed in figures 31 and 32:

1	0.000000000	192.168.137.1	192.168.137.4	H264	56	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=0, Time=0 NAL unit - Access unit delimiter
2	0.000012000	192.168.137.1	192.168.137.5	H264	56	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=0, Time=0 NAL unit - Access unit delimiter
3	0.000081000	192.168.137.1	192.168.137.5	H264	78	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=1, Time=0 NAL unit - Sequence parameter set
4	0.000090000	192.168.137.1	192.168.137.4	H264	78	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=1, Time=0 NAL unit - Sequence parameter set
5	0.000123000	192.168.137.1	192.168.137.5	H264	58	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=2, Time=0 NAL unit - Picture parameter set
6	0.000130000	192.168.137.1	192.168.137.4	H264	58	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=2, Time=0 NAL unit - Picture parameter set
7	0.000169000	192.168.137.1	192.168.137.4	H264	727	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=3, Time=0 NAL unit - Supplemental enhancement information (SE
8	0.000180000	192.168.137.1	192.168.137.5	H264	727	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=3, Time=0 NAL unit - Supplemental enhancement information (SE
9	0.000208000	192.168.137.1	192.168.137.4	H264	429	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=4, Time=0 NAL unit - Coded slice of an IDR picture
10	0.000227000	192.168.137.1	192.168.137.5	H264	358	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=4, Time=0 NAL unit - Coded slice of an IDR picture
11	0.000273000	192.168.137.1	192.168.137.5	H264	523	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=5, Time=0 NAL unit - Coded slice of an IDR picture
12	0.000326000	192.168.137.1	192.168.137.4	H264	310	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=5, Time=0 NAL unit - Coded slice of an IDR picture

Figure 31. GStreamer SPS, PPS, SEI and IDR frames

27	0.045715000	192.168.137.1	192.168.137.5	H264	56	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=8, Time=3600 NAL unit - Access unit delimiter
28	0.045744000	192.168.137.1	192.168.137.4	H264	56	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=8, Time=3600 NAL unit - Access unit delimiter
29	0.045878000	192.168.137.1	192.168.137.4	H264	68	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=9, Time=3600 NAL unit - Coded slice of a non-IDR picture
30	0.045900000	192.168.137.1	192.168.137.5	H264	68	PT=DynamicRTP-Type-96, SSRC=0x7F1D, Seq=9, Time=3600 NAL unit - Coded slice of a non-IDR picture

Figure 32. GStreamer non-IDR frames

An IDR frame is a special type of **I-frame** in H.264. Just to remark that an **I-frame** is an 'Intra-coded picture', in effect a fully specified picture, like a conventional static image file.

P-frames and B-frames hold only part of the image information, so they need less space to store than an I-frame and thus improve video compression rates. An IDR frame specifies that no frame after the IDR frame can reference any frame before it. This makes seeking the H.264 file easier and more responsive in the player.

In the H.264 bitmap it is possible to verify that IDR packet contains an Intra frame and non-IDR packets contains P and B frames.

```
▽ H.264
  ▽ NAL unit header or first byte of the payload
      0... .... = F bit: No bit errors or other syntax violations
      .11. .... = Nal_ref_idc (NRI): 3
      ...0 0101 = Type: NAL unit - Coded slice of an IDR picture (5)
  ▽ H264 bitstream
      0000 0000  0101 1010  001. .... = first_mb_in_slice: 720
      ...0 0010  00.. .... = slice_type: I (I slice) (7)
      ..1. .... = pic_parameter_set_id: 0
      [Not decoded yet]
```

Figure 33. H.264 bitmap of IDR frame

The P frames and B frames are captured and are inside the non-IDR frames.

Just to remember a **P-frame** ('Predicted picture') holds only the changes in the image from the previous frame. For example, in a scene where a car moves across a stationary background, only the car's movements need to be encoded. The encoder does not need to store the unchanging background pixels in the P-frame, thus saving space. P-frames are also known as *delta-frames*.

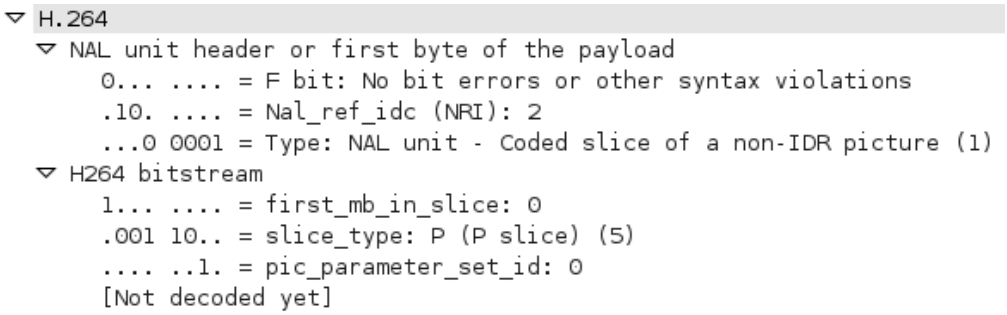

H.264
NAL unit header or first byte of the payload
0... .... = F bit: No bit errors or other syntax violations
.10. .... = Nal_ref_idc (NRI): 2
...0 0001 = Type: NAL unit - Coded slice of a non-IDR picture (1)
H264 bitstream
1... .... = first_mb_in_slice: 0
.001 10.. = slice_type: P (P slice) (5)
.... ..1. = pic_parameter_set_id: 0
[Not decoded yet]

Figure 34. H.264 bitmap of non-IDR frames (P)

A **B-frame** ('Bi-predictive picture') saves even more space by using differences between the current frame and both the preceding and following frames to specify its content.

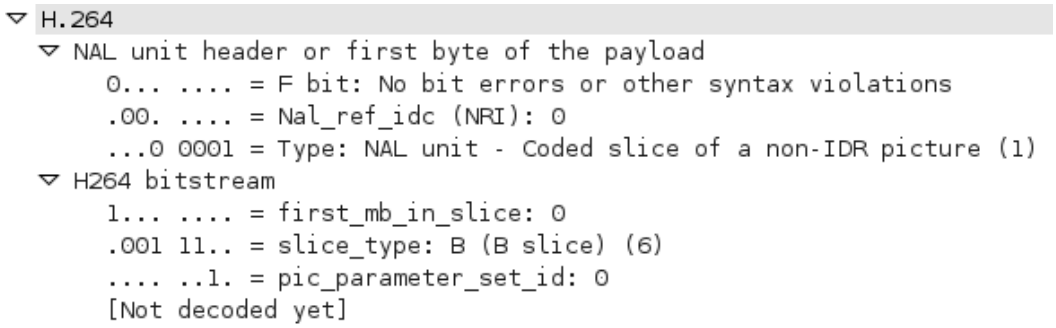

H.264
NAL unit header or first byte of the payload
0... .... = F bit: No bit errors or other syntax violations
.00. .... = Nal_ref_idc (NRI): 0
...0 0001 = Type: NAL unit - Coded slice of a non-IDR picture (1)
H264 bitstream
1... .... = first_mb_in_slice: 0
.001 11.. = slice_type: B (B slice) (6)
.... ..1. = pic_parameter_set_id: 0
[Not decoded yet]

Figure 35. H.264 bitmap of non-IDR frames (B)

I, P and B frames are repeated along the streaming and are seen periodically.

Furthermore, a NAL Unit can be fragmented into different Fragmentation Units (FU-A[12] or FU-B). The structure of FU-B is the same as the structure of FU-A, except for an additional field. The NAL Unit conformed by different FUs is a sequence that can be decoded into video frames. A set of different NAL Units can form a video frame.

In the case of the capture below, 2 different NAL Units are sent (one for each Raspberry) and then this NAL Unit is divided in different FU-As. To indicate the last FU-A packet of a specific NAL Unit, this NAL Unit is marked. In the capture, 2 different Mark FU-A packets that indicate the end of a NAL Unit are shown.

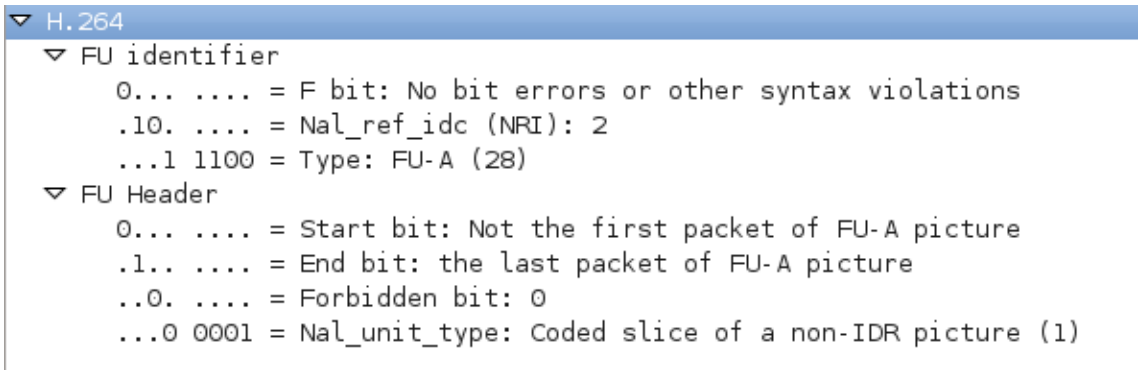

H.264
FU identifier
0... .... = F bit: No bit errors or other syntax violations
.10. .... = Nal_ref_idc (NRI): 2
...1 1100 = Type: FU-A (28)
FU Header
0... .... = Start bit: Not the first packet of FU-A picture
.1.. .... = End bit: the last packet of FU-A picture
..0. .... = Forbidden bit: 0
...0 0001 = Nal_unit_type: Coded slice of a non-IDR picture (1)

Figure 36. FU-A Marked packet

The difference between 2 Marked FU-A packets will provide us the time difference between two equal video frames (one frame for Raspberry A and one frame for Raspberry B).

For instance, the difference between the packets 711 and 712 is **0.046 ms**, which is really small. Fluctiations in this time difference appears. Sometimes the value is lower and sometimes higher but the maximum difference that has been appreciated in the different Wireshark captures performed were about **0.4 ms.**

The video used to provide the WireShark capture has a frame rate of 24 frames per second. This means that time between two consecutive frames is **41.6 ms**.

Having a time difference between packets 711 and 712 of 41.6 ms or higher would mean a decompensation of one or more frames between the 2 video flows. Then, 0.046 ms is a really good value.

As a consequence, the human eye is not able to appreciate time delays between the video flows when seeing them on the monitor screens.

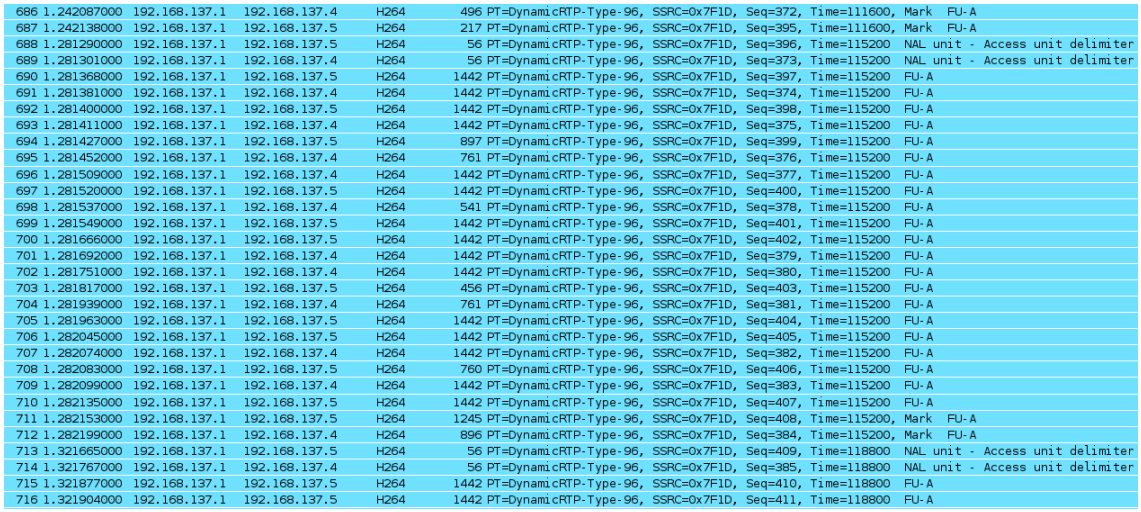


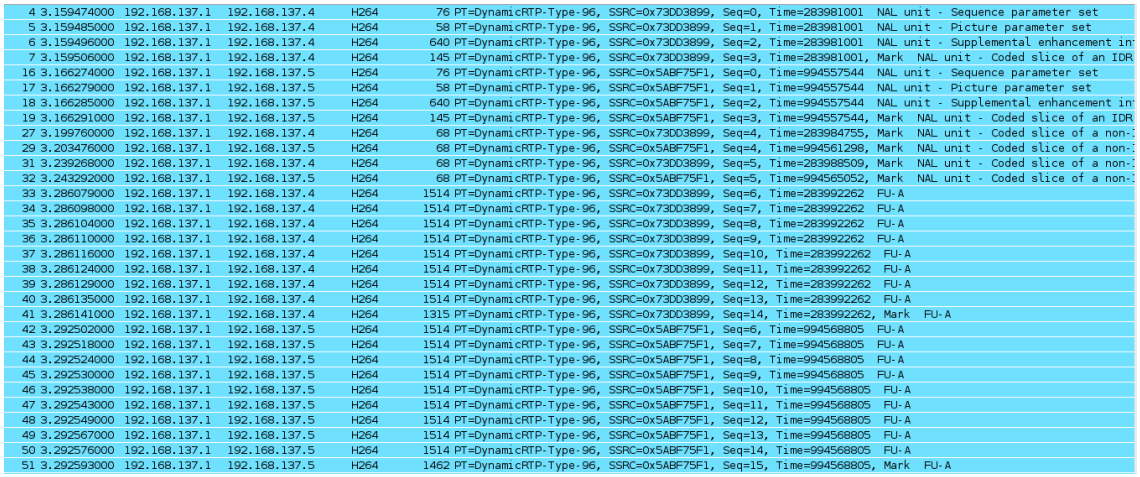
Figure 37. H.264 video flows packets

Finally, while visioning the Video Wall, it is not possible to appreciate any difference in time between both monitor screens. Also, the video was left playing in loop during 5 hours and no synchronization error was appreciated.

5.2. FFmpeg

This section explains the basis of the packets captured with Wireshark during the FFmpeg transmission and why we have synchronization in our Video Wall when using it.

The aim now is to explain the differences between both frameworks. As it can be observed in the figure below, for each video flow a SPS, PPS[13] and SEI NAL units are sent. The functionality is the same as the explained in the previous section.



4	3.159474000	192.168.137.1	192.168.137.4	H264	76	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=0, Time=283981001 NAL unit - Sequence parameter set
5	3.159485000	192.168.137.1	192.168.137.4	H264	58	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=1, Time=283981001 NAL unit - Picture parameter set
6	3.159496000	192.168.137.1	192.168.137.4	H264	640	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=2, Time=283981001 NAL unit - Supplemental enhancement in
7	3.159506000	192.168.137.1	192.168.137.4	H264	145	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=3, Time=283981001, Mark NAL unit - Coded slice of an IDR
16	3.166274000	192.168.137.1	192.168.137.5	H264	76	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=0, Time=994557544 NAL unit - Sequence parameter set
17	3.166279000	192.168.137.1	192.168.137.5	H264	58	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=1, Time=994557544 NAL unit - Picture parameter set
18	3.166285000	192.168.137.1	192.168.137.5	H264	640	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=2, Time=994557544 NAL unit - Supplemental enhancement in
19	3.166291000	192.168.137.1	192.168.137.5	H264	145	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=3, Time=994557544, Mark NAL unit - Coded slice of an IDR
27	3.199760000	192.168.137.1	192.168.137.4	H264	68	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=4, Time=283984755, Mark NAL unit - Coded slice of a non-
29	3.203476000	192.168.137.1	192.168.137.5	H264	68	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=4, Time=994561298, Mark NAL unit - Coded slice of a non-
31	3.239268000	192.168.137.1	192.168.137.4	H264	68	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=5, Time=283988509, Mark NAL unit - Coded slice of a non-
32	3.243292000	192.168.137.1	192.168.137.5	H264	68	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=5, Time=994565052, Mark NAL unit - Coded slice of a non-
33	3.286079000	192.168.137.1	192.168.137.4	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=6, Time=283992262 FU-A
34	3.286098000	192.168.137.1	192.168.137.4	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=7, Time=283992262 FU-A
35	3.286104000	192.168.137.1	192.168.137.4	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=8, Time=283992262 FU-A
36	3.286110000	192.168.137.1	192.168.137.4	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=9, Time=283992262 FU-A
37	3.286116000	192.168.137.1	192.168.137.4	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=10, Time=283992262 FU-A
38	3.286124000	192.168.137.1	192.168.137.4	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=11, Time=283992262 FU-A
39	3.286129000	192.168.137.1	192.168.137.4	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=12, Time=283992262 FU-A
40	3.286135000	192.168.137.1	192.168.137.4	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=13, Time=283992262 FU-A
41	3.286141000	192.168.137.1	192.168.137.4	H264	1315	PT=DynamicRTP-Type-96, SSRC=0x73DD3899, Seq=14, Time=283992262, Mark FU-A
42	3.292502000	192.168.137.1	192.168.137.5	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=6, Time=994568805 FU-A
43	3.292518000	192.168.137.1	192.168.137.5	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=7, Time=994568805 FU-A
44	3.292524000	192.168.137.1	192.168.137.5	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=8, Time=994568805 FU-A
45	3.292530000	192.168.137.1	192.168.137.5	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=9, Time=994568805 FU-A
46	3.292538000	192.168.137.1	192.168.137.5	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=10, Time=994568805 FU-A
47	3.292543000	192.168.137.1	192.168.137.5	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=11, Time=994568805 FU-A
48	3.292549000	192.168.137.1	192.168.137.5	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=12, Time=994568805 FU-A
49	3.292567000	192.168.137.1	192.168.137.5	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=13, Time=994568805 FU-A
50	3.292576000	192.168.137.1	192.168.137.5	H264	1514	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=14, Time=994568805 FU-A
51	3.292593000	192.168.137.1	192.168.137.5	H264	1462	PT=DynamicRTP-Type-96, SSRC=0x5ABF75F1, Seq=15, Time=994568805, Mark FU-A

Figure 38. FFmpeg Wireshark Capture

Furthermore, for each flow, an Intra frame is sent and following this, one P frame and one B frame. I, P and B frames are repeated along the streaming and they are seen periodically.

After that a NAL unit is fragmented in different FU in the two video flows. Here a difference in comparison with GStreamer is appreciated: frames are sent in burst. In this example, for Raspberry A, a set of 9 packets are transmitted and then a set of 10 packets for Raspberry B.

The difference between 2 Marked FU-A packets provides the time difference between two equal video frames (one frame for Raspberry A and one frame for Raspberry B).

In this case the difference between the packets 40 and 51 is **6.452 ms**. It is higher compared with the GStreamer case and there are fluctuations. This test was done 15 times, being the maximum value **7.2 ms**, and the minimum **2.5 ms**.

The same test as in GStreamer was carried out. The video was left playing during 5 hours and after that time it seemed to be synchronized.

CHAPTER 6. CPU USAGE COMPARISON

CPU usage is a term used to describe how much the processor is working. A computer's CPU usage can vary depending on the types of tasks that are being performed by the processor.

CPU usage can be monitored to see how much of the processor's capacity is in use. Excessive CPU usage for certain programs can be a cause for concern, because the server or the Raspberry Pi device may overheat and stop working correctly.

For this reason, it is interesting to know what is the CPU usage for each one of the milestones described in chapter 4. Some measures have been done using top commands. The results are shown below.

6.1. GStreamer

Server CPU Usage

After measuring average CPU usage in the server, the following results are obtained:

The CPU usage for the Single Video Streaming is the lowest because there is no need to decode and re-encode. The video stream is simply encapsulated into RTP packets and the RTP flow is sent via UDP.

The second lowest CPU usage is for the webcam because the webcam resolution is 640x480 and it is easier to encode and send it compared to a 1920x1080 resolution.

The third lowest CPU usage is Local Playback. It decodes the video and decoding process takes a lot of CPU usage.

The fourth lowest CPU usage is for the Single Video Streaming with MPEG2-TS encapsulation. In this case, then decode and re-encode is needed and the element mpegtsmux also adds a delay in the process and needs more CPU.

Finally, the maximum consumption is for the 2 Video Flow Streaming because, in this case, the process of decoding and re-encoding is done twice for two different flows.

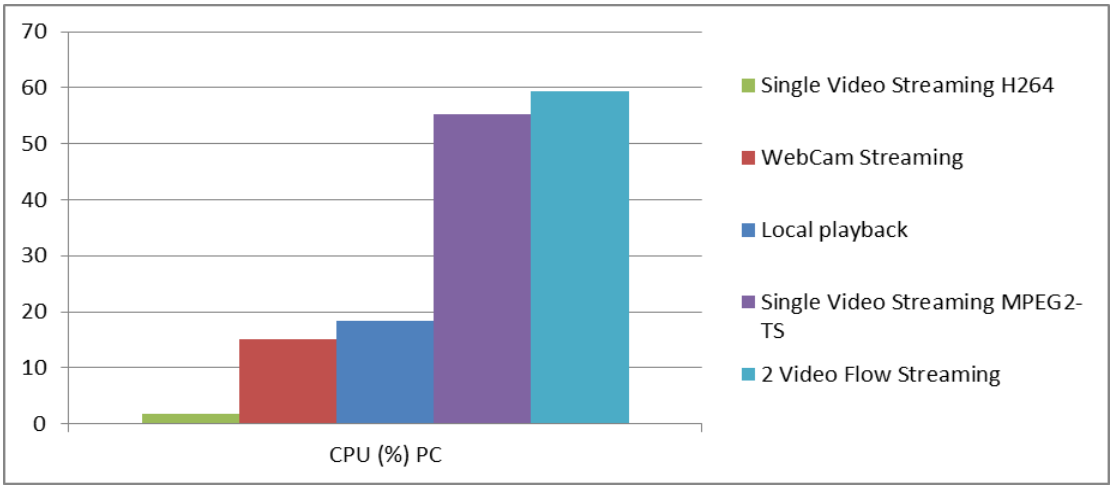


Figure 39. Server CPU Usage Comparison

After measuring average CPU usage in Client 1 and 2, the following results are obtained:

Client 1 CPU Usage

The CPU usage for the WebCam Streaming is the lowest because only 640x480 pixels resolution is being processed.

The second lowest CPU usage is for 2 Video Flow Streaming because the Raspberry only receives a quarter part of the original 1920x1080p (that is 960x540p).

The third lowest CPU usage is Local Playback because the Raspberry does not need to process any stream (no depayloader, no jitter buffer, etc.).

The fourth and fifth lowest CPU usage are for the Single Video Streaming with H.264 and for the Single Video Streaming with MPEG2-TS encapsulation. Single video streaming contains a 1920x1080p video flow and this is harder to decode than any of the previous flows.

The highest CPU usage is for the reception cropping. It takes around 20% increase in CPU usage than the fourth case. The advantage is that the server CPU usage is completely the same (around 2%) whether cropping in reception or sending a Full HD video directly.

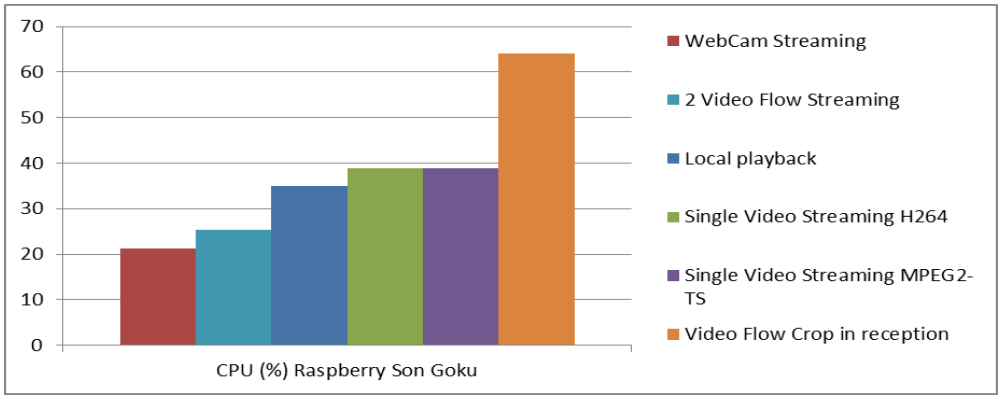


Figure 40. Raspberry 1 CPU Server Usage Comparison

Client 2 CPU Usage

More or less, same results are obtained in Raspberry 2. The CPU usage is between 3-7% more than the first Raspberry. This could be due to hardware specifications are not the same in both Raspberry Pi devices.

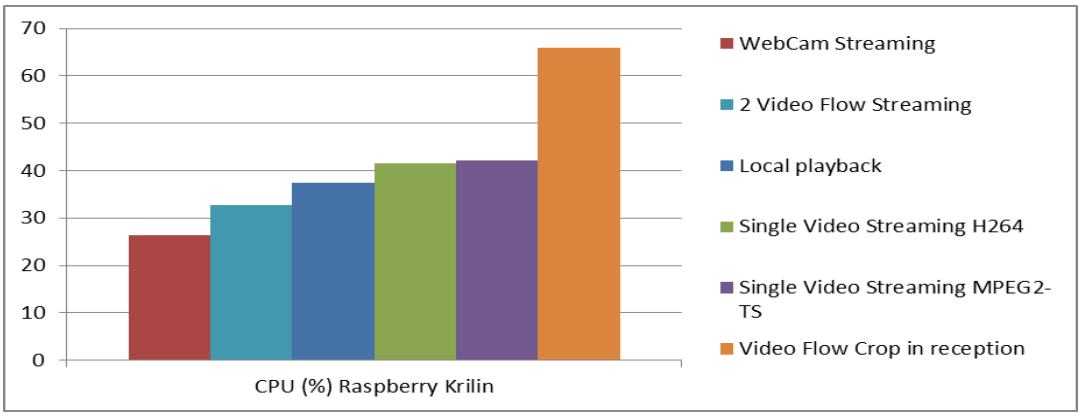


Figure 41. Raspberry 2 CPU Server Usage Comparison

6.2. FFmpeg

Server CPU Usage

After measuring average CPU usage in the server, the following results are obtained:

A single Video Streaming at 720p has the lowest CPU usage (around 24 %).

The second lowest CPU usage is for the Single Video Streaming at 1080p with a CPU usage around 25 %. There is a very low difference in CPU usage but this can be due to Streaming 1080p flow does not work perfectly with FFmpeg.

Finally the Two video flows streaming at 720p have the highest CPU consumption (around 27%). The process of duplicating the streams, cropping and encoding increases a bit the CPU usage even the resolution of both video flows is reduced to a quarter part of the original.

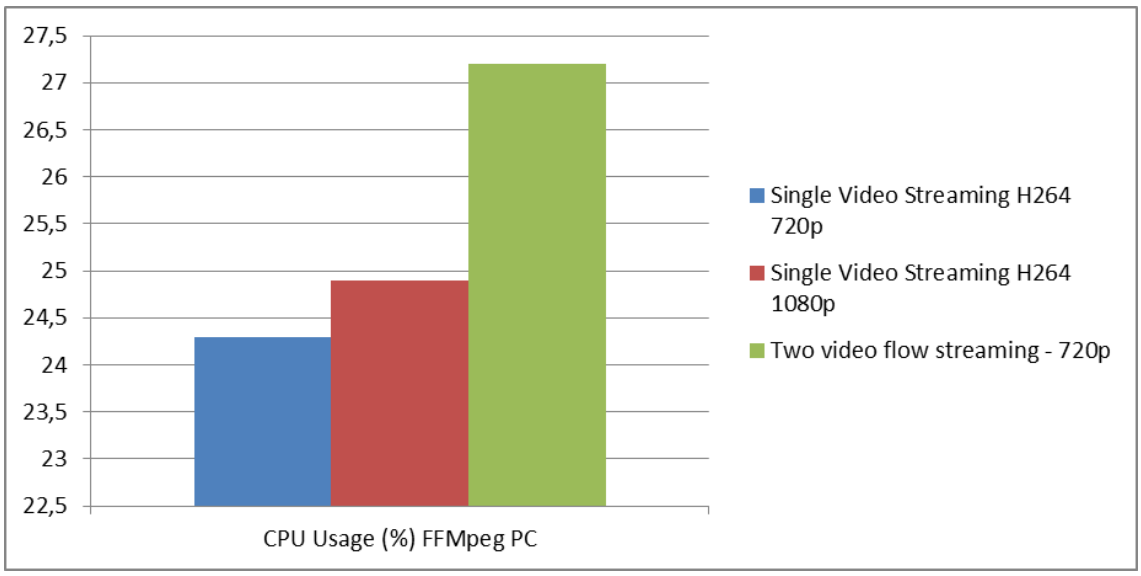


Figure 42. CPU Usage FFmpeg Server

After measuring average CPU usage in Client 1 and 2, the following results are obtained:

Client 1 CPU Usage

The results are more or less the same obtained with GStreamer. The lower CPU usage is for the two video flows streaming. The Raspberry decodes a video with a low resolution and bit rate.

The second lowest CPU usage is for the single video Streaming with a consumption of 37 %. This time, Raspberry is decoding a video with a resolution of 1280x720p.

Finally, the highest CPU consumption is for the 1080p single video streaming. The difference between both is very small and this could be due to the malfunctioning of the FFmpeg pipeline in the server.

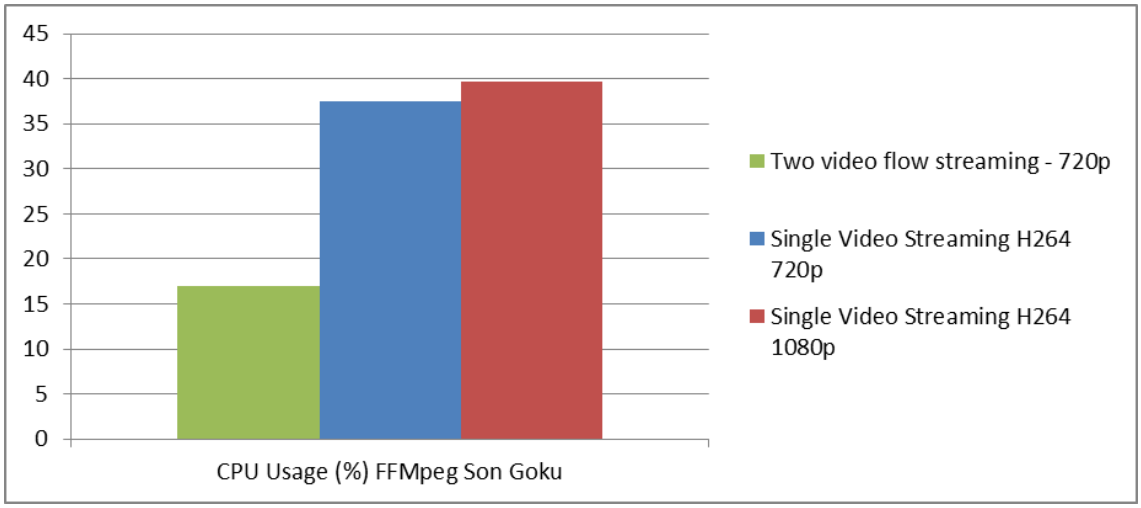


Figure 43. CPU Usage FFmpeg Raspberry 1

Client 2 CPU Usage

Something similar happens with the second Raspberry. Just note that Raspberry 2 CPU usage is always higher than Raspberry 1.

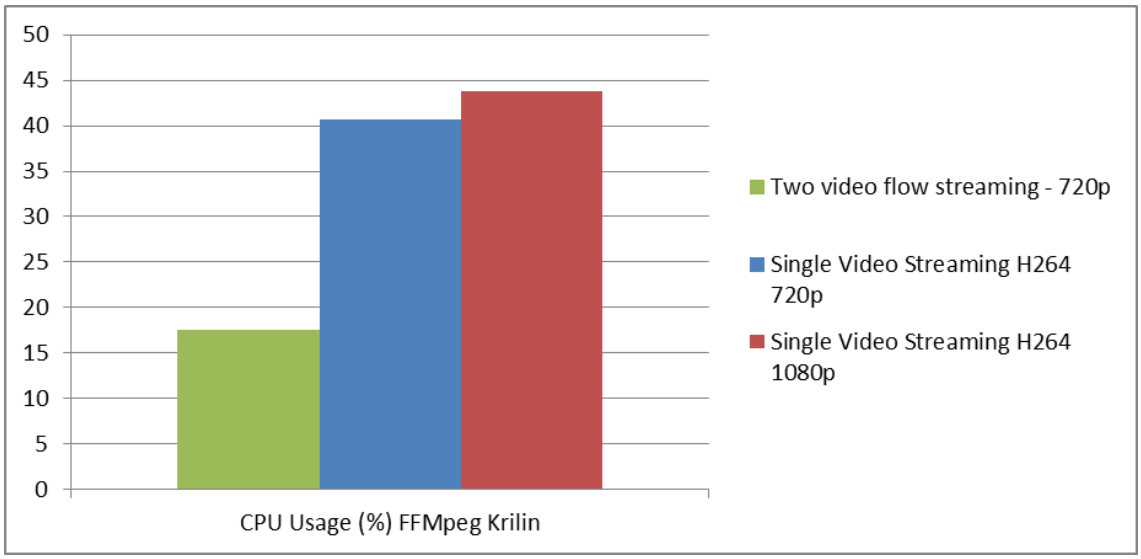


Figure 44. CPU Usage FFmpeg Raspberry 2

6.3. VLC

Server CPU Usage

Using VLC the video flows can be directly streamed even if the video flow uses MPEG2-TS container. In this case, the CPU usage for the Single Video Streaming is the lowest. The video stream is simply payloaded into RTP packets and the RTP flow is sent via UDP.

The CPU usage for the Single Video Streaming with MPEG2-TS encapsulation increases a bit. The increase in CPU usage is due to the muxer. Decode and re encoding, like in GStreamer, is not needed. For this reason, the CPU usage in server decreases (from 55% to just 2.3%).

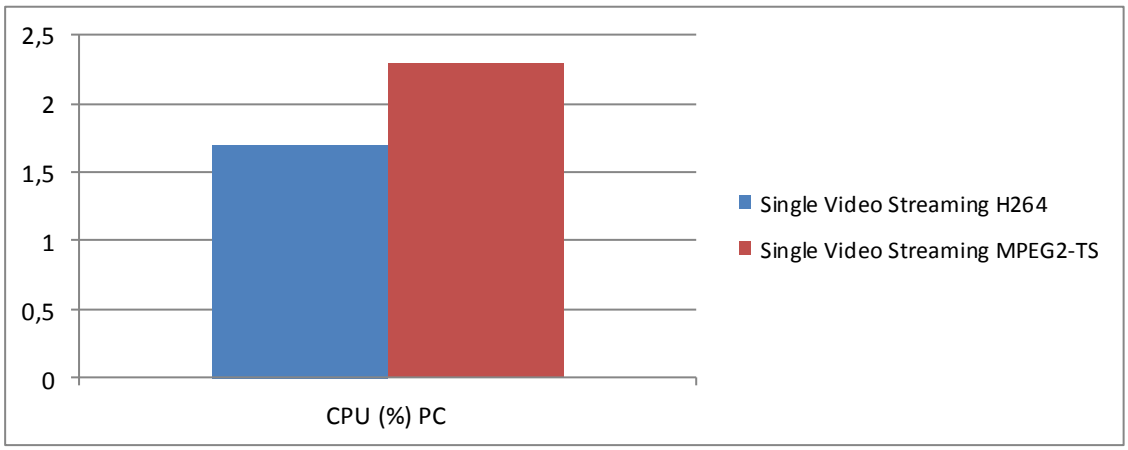


Figure 45. VLC Server CPU Usage

Client 1 CPU Usage

The behaviour of Client 1 and Client 2 is more or less the same. Single Video Streaming with H.264 and Single Video Streaming with MPEG2-TS encapsulation uses more or less the same CPU in reception.

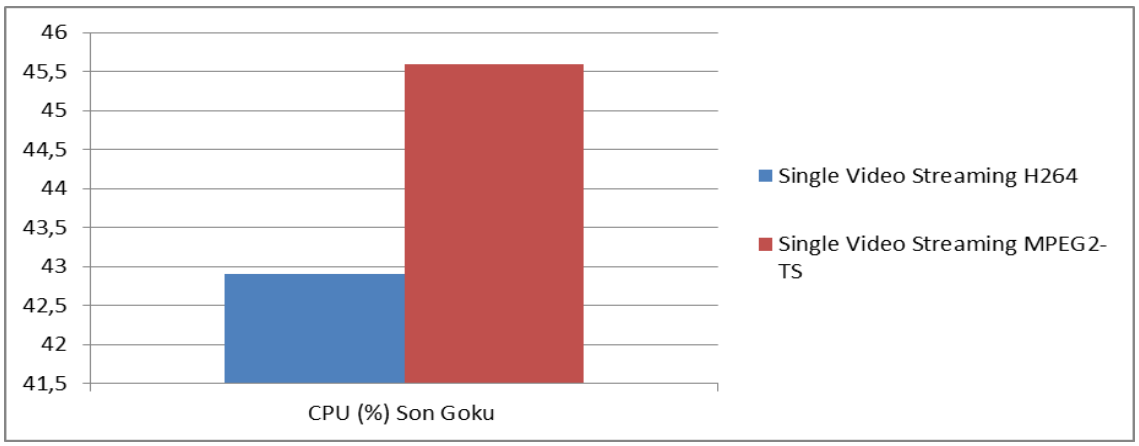


Figure 46. VLC CPU usage Client 1

This also happened when the emmiter was GStreamer. The RTP depayload process causes a bit higher CPU usage in the case of MPEG2-TS container.

Client 2 CPU Usage

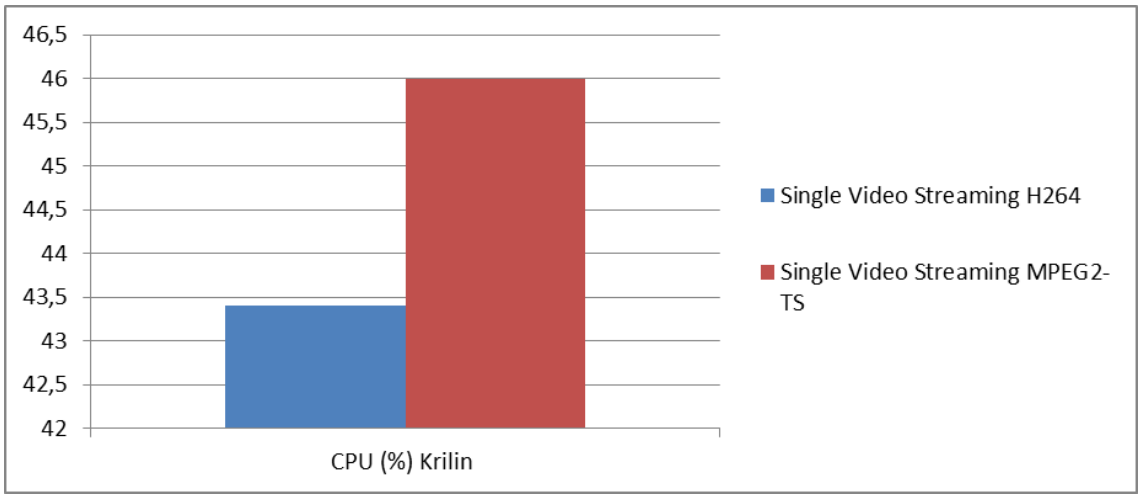


Figure 47. VLC CPU Usage Client 2

Quite similar CPU usage for Raspberry 2 compared to the first Raspberry Pi.

The difference in both graphs, if we compare them to GStreamer, is that videos have a resultion of 1280x720 meanwhile in GStreamer videos had a resolution of 1920x1080.

That means that CPU usage is higher when VLC is used as emmiter than when GStreamer becomes the emmiter. This can be due the differences encountered in the packets transmitted.

6.4. OMXPlayer

Some measures using top command have been done and the results obtained are the following:

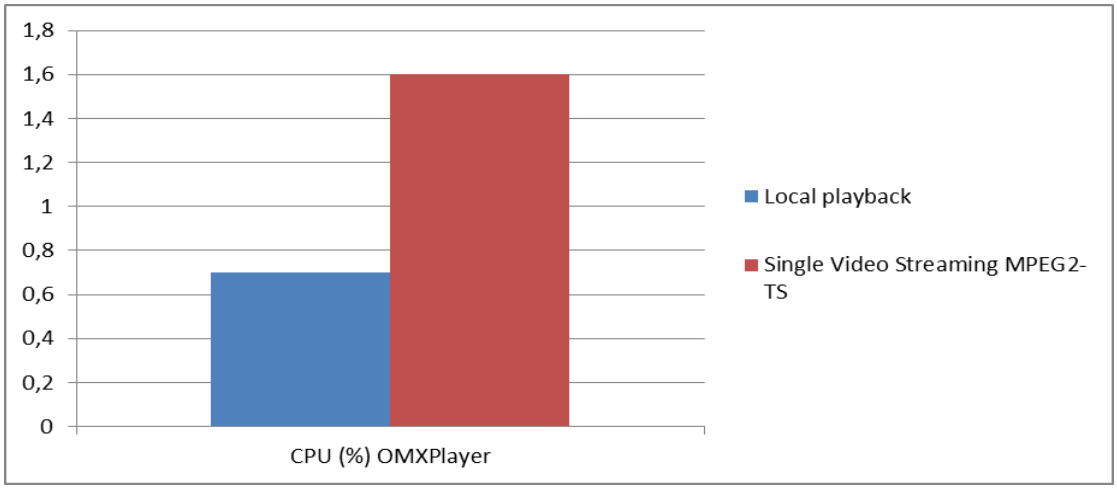


Figure 48. OMXPlayer CPU usage

Both Raspberry Pi devices have similar CPU usage when OMXPlayer is executed. It is for that reason that there is just one figure for CPU usage.

Surprisingly, CPU consumption is 0.7% when we play back a Full HD video. This has nothing to do with GStreamer CPU usage when playing back a local video (it was around 35%). Performance is much better using OMXPlayer than GStreamer and in this case the results are the same.

For the single video streaming using MPEG-2TS container, the CPU usage is doubled (1,6 %) and as commented in OMXPlayer milestones in chapter 4, the results were not satisfactory. However, if we compare it with GStreamer CPU Usage results, once more, the difference is incredible: CPU usage in GStreamer was about 38%.

In conclusion, and in terms of performance, OMXPlayer is better than GStreamer, though, in general, better results have been achieved with GStreamer.

CHAPTER 7. NETWORK TESTING

This chapter will explain the tests performed in the system network used. To perform the testing a program called DummyNet[45] has been used. DummyNet is a live network emulation tool, originally designed for testing networking protocols, and since then, used for a variety of applications including bandwidth management.

This testing allows us to know the behaviour of the video wall in a non-ideal network, by measuring the response of the system against network losses and delays.

7.1 DummyNet

DummyNet simulates/enforces queue and bandwidth limitations, delays, packet losses, and multipath effects. It also implements various scheduling algorithms. DummyNet can be used on the machine running the user's application, or on external boxes acting as routers or bridges.

This tool runs within different operating systems (FreeBSD, OSX, Linux, Windows) and works by intercepting selected traffic on its way through the network stack, as in the figure below, and passing packets to objects called **pipes** which implement a set of queues, a scheduler, and a link, all with configurable features (bandwidth, delay, loss rate, queue size, scheduling policy...).

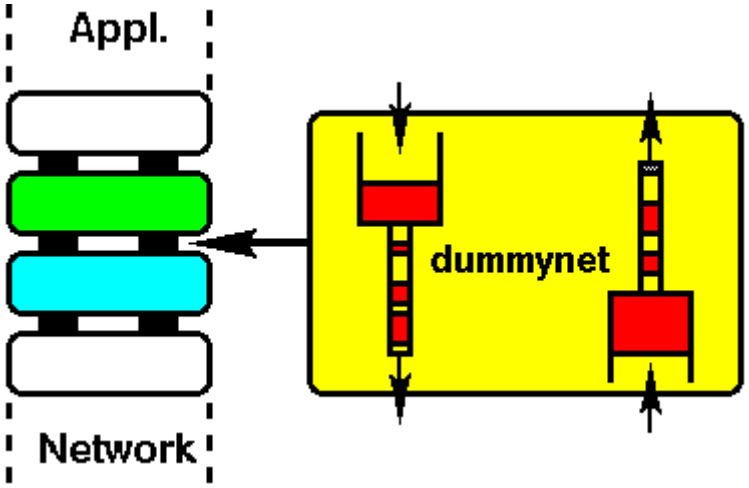


Figure 49. DummyNet Operation

Traffic selection is done using the **ipfw** firewall, which is the main user interface for dummynet. Ipfw allows to select precisely the traffic and direction desired to work on. Multiple pipes can also be created, traffic can be sent to different pipes, and even cascades of pipes be built.

7.2. DummyNet Scenario

Our scenario is very similar to the scenario in the Figure 26, though there is a new element between the Server and the Router ADSL.

This new element is a PC running Frenzy. This is a portable system administrator toolkit, LiveCD based on FreeBSD. It generally contains software for hardware tests, file system check, security check, network setup and analysis. DummyNet is a standard component in FreeBSD.

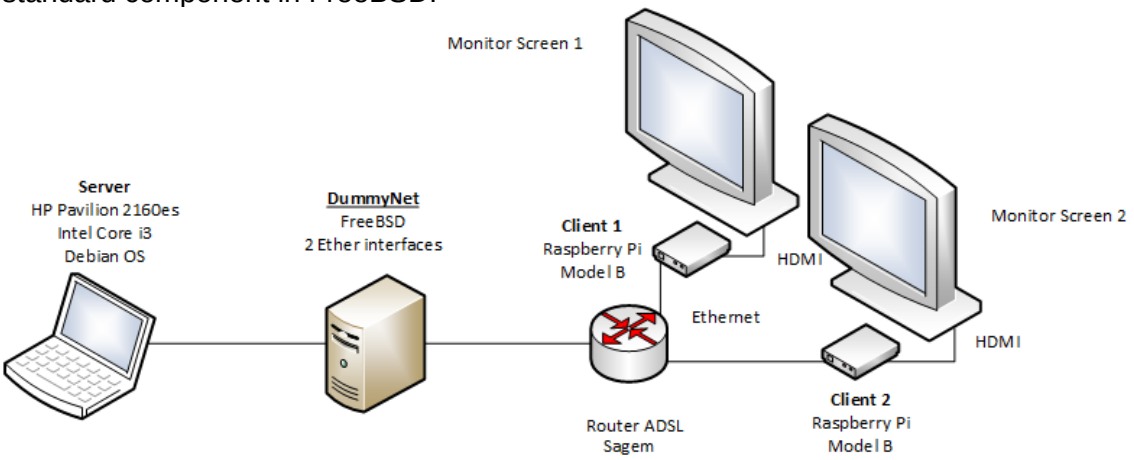


Figure 50. DummyNet Testing Scenario

This machine running DummyNet has 2 network interfaces because the traffic between the Server and the two Raspberry Pi devices has to be intercepted to add delay and losses. This test will permit to know the behaviour of the video wall in a non-ideal network.

7.3. DummyNet Configuration

First, IP address and netmask are set to both interfaces.

The Server is connected to the **em0** interface and the Router ADSL to the **bge0** interface. The address and netmask are set with the **ifconfig** command.

```
ifconfig bge0 192.168.137.3 netmask 255.255.255.0
ifconfig em0 192.168.137.2 netmask 255.255.255.0
```

A bridge is created to interconnect **bge0** and **em0**. The basic operation of a bridge is to join two or more network segments together. The bridge works by learning the MAC layer addresses (Ethernet addresses) of the devices on each of its network interfaces. It forwards traffic between two networks only when its source and destination are on different networks.

```
ifconfig bridge create
```

This should return **bridge0** if this is the first bridge we create. The two interfaces used in the bridge will be used: **bge0** and **em0**.

```
ifconfig bridge0 addm em0 addm bge0 up
```

Now the command ifconfig-up is executed to the bridge and the two interfaces.

```
ifconfig em0 up
ifconfig bge0 up
ifconfig bridge0 up
```

The gateway list has to be verified too, and the gateways, added, to achieve the endpoints (Server and 2 Raspberry Pi devices).

```
netstat -rn
route add 192.168.137.5 192.168.137.3
route add 192.168.137.4 192.168.137.3
route add 192.168.137.1 192.168.137.2
```

7.4. Performed test and results

Once there is end-to-end connectivity in the new scenario, a pipe in DummyNet is created to apply delay and losses[46].

The first step is to check whether there are any active rules that can block the traffic. These rules are deleted using the **flush** parameter.

```
ipfw list
ipfw -q -f flush
```

A pipe is created to test the scenario. This command will add the **pipe 1** that permits to add delays and losses to the packets from any to any IP address in the scenario.

```
ipfw add 100 pipe 1 ip from any to any
```

Then a delay that will affect both video flows is applied. This command will apply a delay of 100ms.

```
ipfw pipe 1 config delay 100ms
```

Losses to both video flows can be applied using the **config plr** parameter. For instance, this command applies a 2% of losses.

```
ipfw pipe 1 config plr 0.02
```

Finally, if needed, **pipe 1** can be deleted by using the following command.

```
ipfw delete 100 pipe 1 ip from any to any
```

The video wall has been tested applying delays and losses to just one Raspberry and applying different values of delays and losses to each one of the Raspberry Pi devices. For these tests two new pipes have to be created and the same as above, applied.

```
ipfw add 100 pipe 2 out dst-ip 192.168.137.4
ipfw add 100 pipe 3 out dst-ip 192.168.137.5
```

The results of these tests are very satisfactory. Our Video Wall can recover perfectly the 2 video flows after applying some delays or losses.

When losses are applied, the image quality decreases a lot because it is a digital communication. If no more losses are added to the network, the Video Wall recovers the streaming perfectly after refreshing the image.

In case of applying delays, a difference in time is appreciated when the audio in the server is heard before watching the image on the 2 monitor screens. If no more delays are added, the Video Wall instantly returns to display the actual image synchronized with the audio on the screen.

After testing this scenario, it is concluded that the system is synchronous and robust since the 2 Raspberry Pi devices perfectly recover the transmission after applying delays and losses to one of the Raspberry Pi device or both. No frame descompensations between the 2 video flows have been observed.

CHAPTER 8. THROUGHPUT AND BACKPLANE ANALYSIS

This chapter will analyse the network throughput and the backplane that will handle the switch/router in the scenario of Figure 26.

Here are the definitions of the two concepts that will be analysed later:

Throughput or **network throughput** is the average rate of *successful* message delivery over a communication channel. This data may be delivered over a physical or logical link, or pass through a certain network node. The throughput is usually measured in bits per seconds (bit/s), and sometimes in data packets per second or data packets per time slot.

Backplane capacity is the shared capacity between interfaces on a switch. Total switch bandwidth can be determined by adding the available port bandwidth. However, the backplane limitations have to be taken into account. A switch is limited by backplane capacity. For instace, a switch of 10 ports at 100 Mbps with a backplane capacity of 500 Mbps will not be able to use all the 10 ports at full capacity.

Normally, 10 Mbps switchs have enough backplane. However, for 100 Mbps and Gigabit Switch the cost is higher when trying to achive enough Backplane.

The throughput analysis for the following milestones defined in chapter 4 will be provided.

8.1. Milestone 9 - Throughput and Backplane

Throughput

In Milestone 9, the pipeline splits the original video flow in the server and sends a quarter of the original video to each of the Raspberry Pi devices.

Capturing with Wireshark, for a single quarter, the following results are obtained:

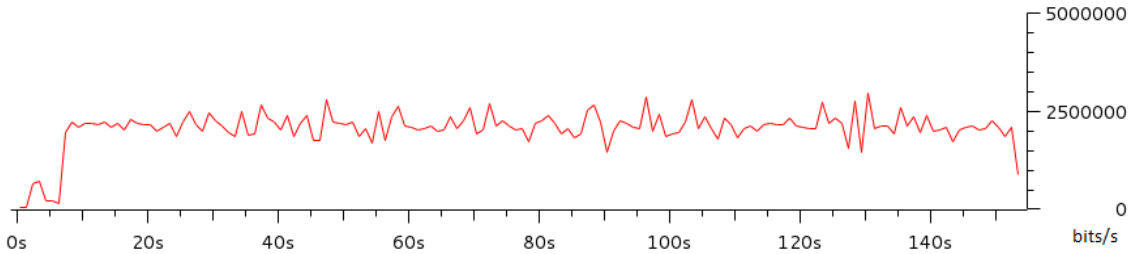


Figure 51. Throughput 960x540 single flow

The throughput is around **2.5 Mbps**. Notice that this throughput should be multiplied by 4 because each of the Raspberry Pi devices should take their own flow.

Therefore, the switch should take each of the flows and forward each of them to 4 different ports.

Backplane

In this case, the backplane capacity of the switch should be:

Backplane capacity = 4 * 2.5 Mbps = **10 Mbps**

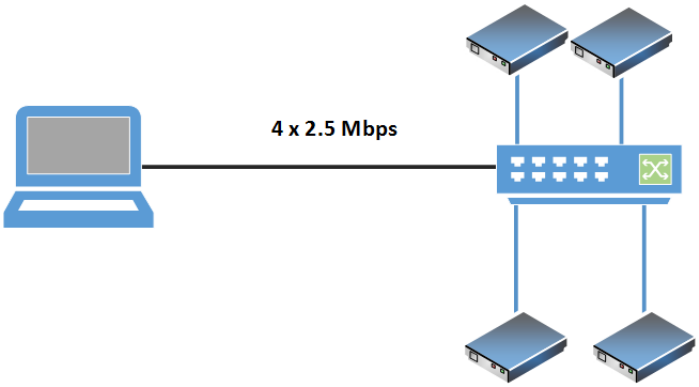


Figure 52. Backplane capacity Milestone 9

8.2. Milestone 11 - Throughput and Backplane

Throughput

In milestone 11, the pipeline sends in broadcast the original video flow. Audio is not included in order to make a comparison with the throughput obtained before:

Capturing with Wireshark, the following results are obtained:

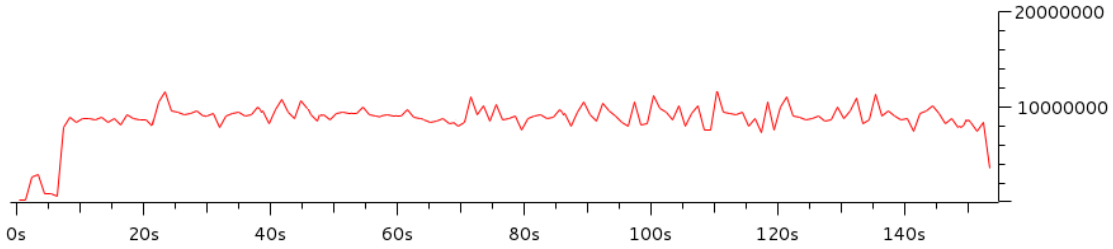


Figure 53. Throughtput 1920x1080 single flow

The throughput is around 10 Mbps as expected. In this case the switch should take this flow and forward directly to the different 4 ports.

Backplane

The Backplane capacity in this case is the following:

Backplane capacity = **10 Mbps**

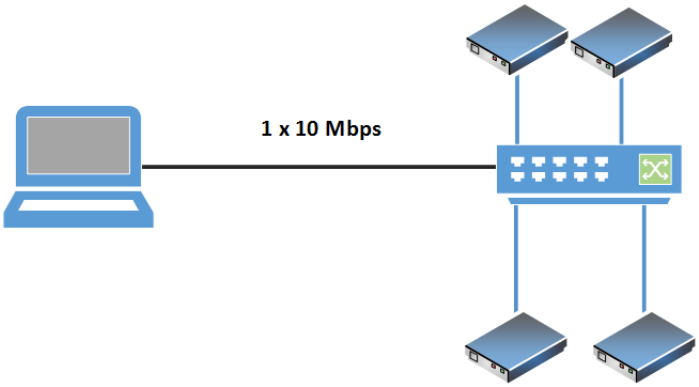


Figure 54. Backplane capacity Milestone 11

A backplane of 10 Mbps is not very high. A wide range of switchs provides this backplane or more for a low price.

In case of having a server with good features it is possible to split the video and then send it through the network. Otherwise, if server is not good enough it is better to use the pipeline defined in Milestone 11.

CHAPTER 9. VIDEO WALL CONTROL SYSTEM

Pipelines have been used all along this project: pipelines for emmiting and pipelines for receiving video flows. A lot of testing has been done. Changes from one pipeline to another, as well as verifying different scenarios.

Obviously, the future user of the implemented system wants the system to be very simple and easy to use. It is not expected that the client will change from one pipeline to another, as the different test cases explained in chapter 4.

Therefore, once the project is finished, how to provide a real case commercial scenario will be the next step.

9.1. Control system description

In this case, it is thought that this project can be built in different small stores since a 2x2 videowall is the best approach for the project due to the quality when splitting. If a Full HD video is divided into 4 screens, quality is still good enough.

The idea is to provide freedom to the person in charge of the content of the VideoWall.

The following figure provides an idea of the scenario:

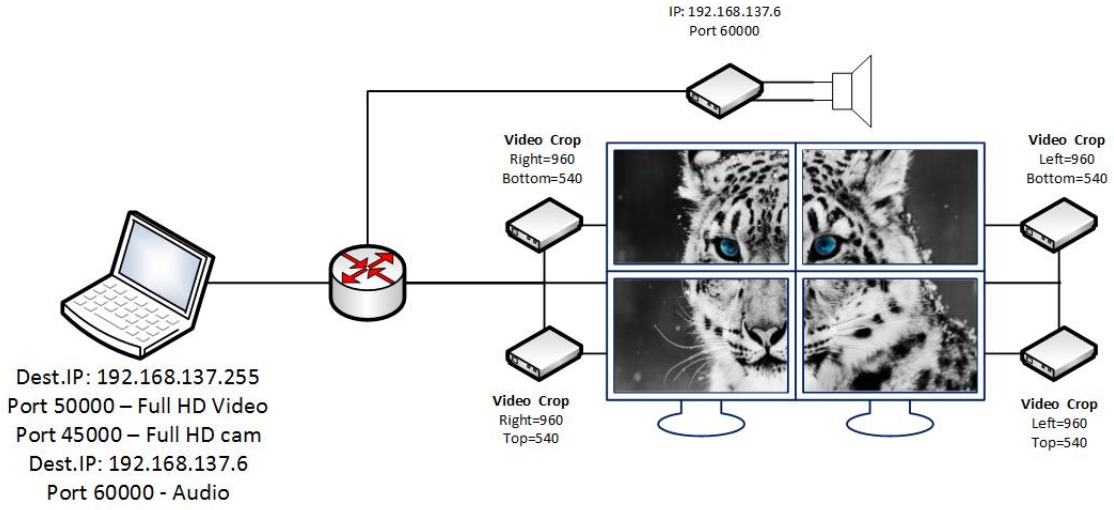


Figure 55. Commercial use scenario

To make it simple, a control system has been implemented. Its functions will be the following:

1. In the Raspberry Pi startup, three different pipelines contained in different Shells scripts (.sh) will be executed:

Pipeline	Description
Information message	Clean the screen and print a welcome message
H.264 & WebCam receiver	Milestone 4 Pipeline (GStreamer)
H.264 & WebCamCropping receiver	Milestone 10 Pipeline (GStreamer)

Table 4. Initial Scripts in Raspberry Startup

Web Cams should be Full HD cameras since the cropping script is designed to split a Full HD video in 4 differents parts.

2. In the Raspberry that will handle the audio stream the following pipeline will start:

Pipeline	Port	Description
H.264 audio receiver	60000	Milestone 8 Pipeline (GStreamer)

Table 5. Initial Script for Audio

The following figure shows a capture of the initial screen of Raspberry Pi devices when the H.264 receiver script and the H.264 cropping script are executed in the start up.

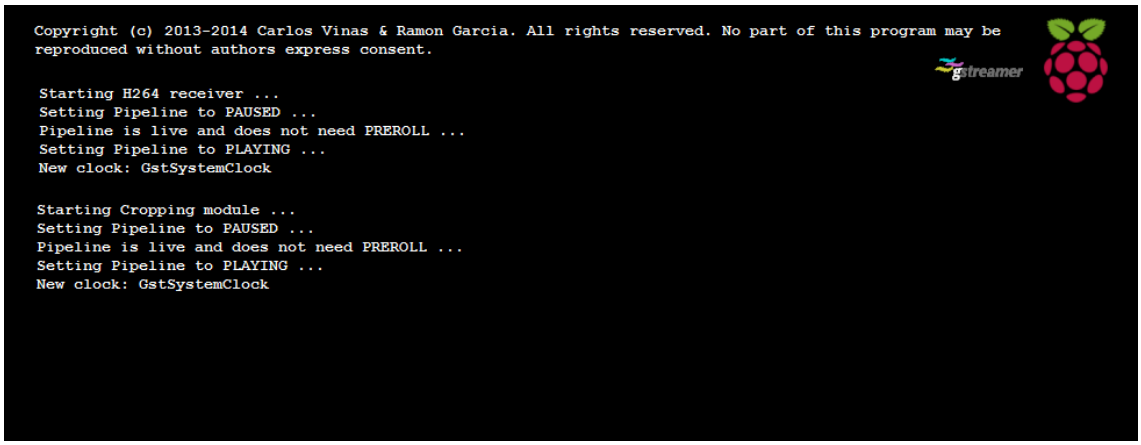

Copyright (c) 2013-2014 Carlos Vinas & Ramon Garcia. All rights reserved. No part of this program may be reproduced without authors express consent.
Starting H264 receiver ...
Setting Pipeline to PAUSED ...
Pipeline is live and does not need PREROLL ...
Setting Pipeline to PLAYING ...
New clock: GstSystemClock
Starting Cropping module ...
Setting Pipeline to PAUSED ...
Pipeline is live and does not need PREROLL ...
Setting Pipeline to PLAYING ...
New clock: GstSystemClock

Figure 56. Start up screen Raspberry Pi devices

3. In the server, the user can switch between 2 different pipelines:

Pipeline	Description
H.264 emmiter (crop and without crop)	Milestone 11 Pipeline (GStreamer)
WebCam emmiter (crop and without crop)	Milestone 3 Pipeline (GStreamer)

Table 6. Scripts in the server

The graphical user interface (GUI) will be the following:

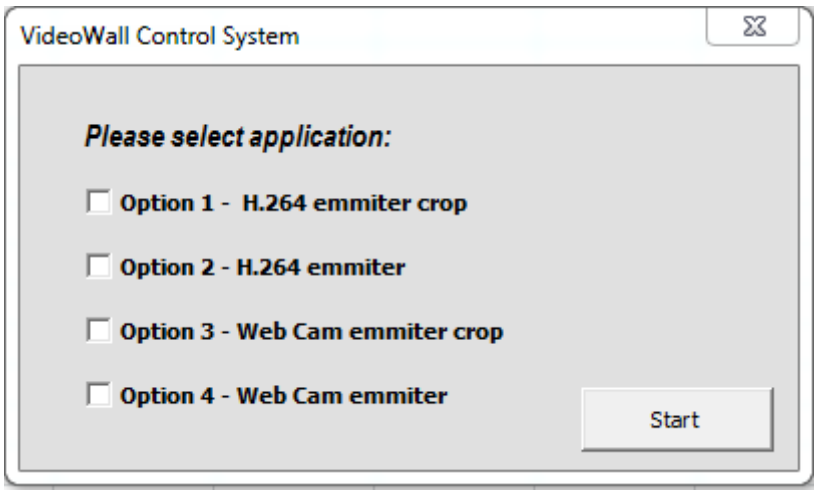

VideoWall Control System
Please select application:
Option 1 - H.264 emmiter crop
Option 2 - H.264 emmiter
Option 3 - Web Cam emmiter crop
Option 4 - Web Cam emmiter
Start

Figure 57. Server VideoWall GUI

The destination IP in the emitter pipelines will be the broadcast address. The port will change depending on the type of source (video flow, video flow cropped, web cam flow or webcam cropped). The user will indicate the preferred option in the shell script in the server as in the following table:

Pipeline	Option	IP Address	Port
H.264 emmiter crop	1	192.168.137.255	45000
H.264 emmiter	2	192.168.137.255	50000
Web cam emmiter crop	3	192.168.137.255	45000
Web cam emmiter	4	192.168.137.255	50000

Table 7. IP and Port configuration

9.2. Startup scripts configuration

This section will explain how to configure the shell scripts that will run in the start up on each one of the Raspberry Pi devices.

The initialization configuration in Linux is done in the following location:

```
/etc/init.d/
```

This folder contains the scripts that will be executed during the startup process. Here the 4 startup scripts (Welcome message, H.264 and Web Cam receiver, H.264 and WebCam crop receiver and Audio Receiver) explained in Section 9.1. will be created.

The scripts must contain the following header (apart from the pipelines) to work correctly:

```
#! /bin/sh

### BEGIN INIT INFO
# Provides:          z01
# Required-Start:    $remote_fs $syslog $network
# Required-Stop:     $remote_fs $syslog $network
# Default-Start:     2 3 4 5
# Default-Stop:      0 1 6
# Short-Description: Simple script to start a program at boot
# Description:       A simple script which will start / stop a
program a boot / shutdown.
### END INIT INFO

# If you want a command to always run, put it here

Here we need to include our pipeline.
```

It is important to name the scripts correctly. All the scripts will be run at the end of the start up. This means, when all the other scripts have been executed.

To do so, the script has to be named with letter z. This will produce that this script will be the last one to be executed since the other ones inside init.d begin with other letters that come before z.

When the script is written, it is made executable. To do so:

```
sudo chmod 755 /etc/init.d/z01.sh
```

Finally, to register the script to be run at start-up and shutdown, the following command is run:

```
sudo update-rc.d z01.sh defaults
```

If ever the script wants to be removed from start-up, the following command will be run:

```
sudo update-rc.d -f  NameOfYourScript remove
```

CHAPTER 10. COMPARISON WITH COMERCIAL PRODUCTS

As explained in chapter 1, there are several Video Wall commercial products. One of the most popular is SAGE.

This chapter will compare the requirements of SAGE with the requirements of the Video Wall in this project to demonstrate that it is a Low Cost Video Wall, in terms of cost, power consumption, backplane and throughput needed.

10.1. SAGE Overview

SAGE (Scalable Adaptive Graphics Environment) is cross-platform, open-source middleware that enables users to have a common operating environment, or framework, to access, display and share a variety of content – whether digital cinema animations, high resolution images, high-definition video-teleconferencing, presentation slides, documents, spreadsheets or laptop screens – in a variety of resolutions and formats, from multiple sources, to one or more tiled display walls.

Figure 58. SAGE working demonstration

SAGE graphics streaming architecture supports the sharing of high-resolution media and data between visualization environments or among groups of collaborators at local or remote sites.

Its network-centric architecture allows collaborators to share visualizations (such as remote desktop, video streams, 2D maps, and 3D renderings) on local or remote systems. Applications stream pixels over high-speed networks to displays with potentially hundreds of megapixels of contiguous display resolution.

SAGE is software for organizing images, animations, and information on scalable display walls (OptIPortals) to help people deal with problems of scale and complexity in their data. It specializes in streaming images from remote rendering servers / supercomputers.

SAGE installations can vary from basic, to intermediate and advanced configurations to support a range of collaborative session requirements. The basic and intermediate configurations focus on local collaboration, with enhanced high-speed desktop sharing and whiteboard functionality, augmenting the intermediate installation. The advanced

configuration supports highly interactive, multi-site remote, collaborative work, and enables multi-user interaction using a variety of input devices: 3D mouse, touch-screen and motion sensing input devices such as the Microsoft Kinect.

10.2. SAGE Features and Tiled Display Wall Configurations

Basic SAGE

A basic SAGE installation only supports local collaborative sessions by allowing multiple participants to share their desktop and assets to a local tiled display or to remote sites. The basic system does not require high-speed network for streaming applications. Desktop content is shared to and controlled on the local display through use of the SAGE Pointer application.

Intermediate SAGE

In addition to the functionality provided by a basic SAGE installation, an intermediate configuration supports improved streaming of the desktop via the SAGE qShare application. It enables desktop applications to be shown on the local tiled display, for instance interactive whiteboard software, video editing suite or 3D modeling software.

Through desktop sharing, third party applications such as Skype can be incorporated into a SAGE session for enhanced communication between participants. It is important to note that third party applications are not integral to SAGE, though are launched as standalone applications that are shared via SAGE.

Advanced SAGE

An advanced SAGE configuration provides functionality for highly interactive local and remote collaborative sessions by sharing and streaming media and data between multiple SAGE-enabled sites. Integration of video conferencing provides for enhanced communication between distributed participants, which is an integral component of networked collaboration. An additional benefit is real-time frame rates provided by the SAGE Capture hardware solution.

Advanced configurations may also incorporate a variety of user interaction alternatives, such as a multi-touch, head tracking, and motion sensing input devices for 3D hand/body tracking. Using EVL's Omicron software development kit (SDK), input data from a number interface devices – such as the SAGE Pointer or commodity game controllers are streamed to the SAGE application for enhanced interactive functionality.

10.3. SAGE Minimum Systems Requirements and Minimun Budget

There are two types of hardware configurations that can support SAGE - a single machine or a cluster based configuration. Though a single machine alternative may be attractive due to its apparent streamlined nature for overall systems administration and price, this alternative is far less scalable beyond a basic tiled environment (limited to 4x4 tiled display). A GPU workstation will still have the capacity to do video capture and network at 10G, but at a cost almost equivalent to a small cluster. Bear in mind, the resolution of the tiled display, stereo capabilities, etc. all factor significantly into the choice of computer infrastructure supporting SAGE. There is seemingly greater flexibility and scalability of using a cluster based approach, especially considering rapidly changing display technologies.

Basic SAGE Minimun Requirements

- 1 control computer (SAGE manager)
- 2 displays/computer (# display tiles to computer ratio)

Specifications/Computer

- Dual Xeon E5-2690 Supermicro GPU server
- NVIDIA GTX680 graphics card (per machine)
- Dual 10Gbps network card (per machine)
- 160GB SSD
- 2TB hard drive

Networking

- 10Gbps network switch

Audio

- Asus Xonar DX PCI Express Sound Card
- 2 speakers (Dell AY410 Multimedia Speaker System) connected to control computer

Intermediate SAGE (Basic SAGE +)

Mac mini

- 2.7GHz Dual-Core Intel Core i7
- 8GB 1333MHz DDR3 SDRAM - 2x4GB
- 256GB Solid State Drive
- Mini Display Port to DVI Adapter

qShare

- Wacom Cintiq 24HD Interactive Pen Display (whiteboard) controlled by the MacMini

Networking

- 1Gbps network connection to the 10Gbps switch running SAGE

Audio

- 4 Channel audio mixer (Rolls MX44s Mini-Mix IV)
- Rolls PS27 12VDC AC adapter

Only taking into account the Basic SAGE minimum requirements, the approximate cost will be:

Element	Cost
2 x Dual Xeon E5-2690 Supermicro GPU server	2 x 1508.78 €
2 x NVIDIA GTX680 graphics card	2 x 380.67 €
2 x Dual 10Gbps network card	2 x 302.34 €
2 x 160GB SSD	2 x 197.66 €
2 x 2TB hard drive	2 x 74.95 €
10Gbps network switch	750.40 €
2 x Monitor Screens	2 x 77.95 €
Asus Xonar DX PCI Express Sound Card	65.88 €
2 x Dell AY410 Multimedia Speaker System	2 x 43.92 €
TOTAL	**5988.82 €**

Table 8. Basic SAGE Cost

10.4. Minimum Systems Requirements and Budget of this project

Chapter 1 described the features of the Server and of the Raspberry Pi devices. We already had some of the elements of the system and the others elements were provided by the i2CAT Foundation.

The following table is an estimated summary of the cost of the components that needed in the Video Wall in case of buying all the elements.

Element	Cost
Server (i3 processor)	389 €
2 x Raspberry Pi Model B - 512 MB	2 x 28.09 €
Router Sagem Fast 2604	15.45 €
2 x Monitor Screens	2 x 77.95 €
2 x HDMI Cables	2 x 2.75 €
3 x Ethernet Cables (Cat 5 E)	3 x 2.50 €
2 x Micro USB Charger	2 x 3.95 €
Speakers	15.75 €
TOTAL	**653 €**

Table 9. Our system cost

The computer used in this project is a notebook that cost 600 € in 2010. Today, it is possible to buy a desktop computer with better specifications and higher performance for the same price. Today, the actual cost of a server with the same features as the one used in this project is 389 €.

In case of building a 2x2 Video Wall, with 5 Raspberry Pi Devices (4 for video an 1 for audio) and the other material needed, the estimated cost is calculated in the table below.

Element	Cost
Server	389 €
5 X Raspberry Pi Model B - 512 MB	5 x 28.09 €
Switch (8 ports 10/100Mbps)	12.75 €
4 x Monitor Screens	4 x 77.95 €
4 x HDMI Cables	4 x 2.75 €
6 x Ethernet Cables (Cat 5 E)	6 x 2.50 €
5 x USB – Micro USB Charger	5 x 3.95 €
Speakers	15.75 €
TOTAL	**915.50 €**

Table 10. 2x2 Project Video Wall cost

Furthermore, to compare the project's Video Wall cost to the cost of SAGE, the following table has been done, ignoring the cost of the cables needed, and using 1 Raspberry Pi with Speakers:

Element	Cost
Server (i3 processor)	389 €
3 x Raspberry Pi Model B - 512 MB	3 x 28.09 €
Switch (8 ports 10/100Mbps)	12.75 €
2 x Monitor Screens	2 x 77.95 €
3 x USB – Micro USB Charger	3 x 3.95 €
Speakers	15.75 €
TOTAL	**669.52 €**

Table 11. Fundamental elements to compare with SAGE

Conclusion

The Basic SAGE costs about **5989 €** and the amount of money needed to build the system of this project's Video Wall costs only **670 €**. That means, **5319 € less than the Basic SAGE.**

10.5. Comparison of the power consumption results

In this section the approximate Power Consumption of the Basic SAGE and this project are calculated.

The Basic SAGE audio system (sound card + 2 speakers) will not be taken into account in order to make a fair comparison to the power consumption of the Video Wall.

The following table shows the approximate power consumption of Basic SAGE. All the elements are considered to be active; therefore, all the elements consume power.

Element	Cost
2 x Dual Xeon E5-2690 Supermicro GPU server	2 x 135 W
2 x NVIDIA GTX680 graphics card	2 x 195 W
2 x Dual 10Gbps network card	2 x 14.32 W
2 x 160GB SSD	2 x 3 W
2 x 2TB hard drive	2 x 6 W
10Gbps network switch	58.8 W
2 x Monitor Screens	2 x 17 W
TOTAL	**799.44 W**

Table 12. Basic SAGE Power Consumption

In case of this project, the estimated maximum power consumption is calculated in the following table (without considering audio):

Element	Cost
Server (i3 processor)	65 W
2 x Raspberry Pi Model B - 512 MB	2 x 3.5 W
Switch (8 ports 10/100Mbps)	2,5 W
2 x Monitor Screens	2 x 17 W
TOTAL	**108.5 W**

Table 13. Basic Power Consumption of our system

Conclusion

Basic SAGE has an approximate power consumption of **799.44 W** and Video Wall only have an estimated power consumption of **108.5 W**. This means that Video Wall consumes around **691 Watts** less than Basic SAGE.

Furthermore, in 2014 the price of KWh in Spain is **0.1752 euros** and 1 kWh = 1000 Wh

In the case of System elaborated in this project, 1000 Wh divided by 108.5 watts = **9.21 hours of uninterrumped work functioning.**

In the case of SAGE System, 1000 Wh divided by 799.44 watts = **1.25 hours of uninterrumped work functioning.**

For the same price, **8 hours more** on uniterrumped work functioning is obtained by the system created along the project.

10.6. Throughput and Backplane analysis

In this section the network throughput and the backplane that will handle the Basic SAGE system are estimated in order to compare them to the results obtained with Low Cost Video Wall.

Throughput analysis

The Basic SAGE needs a 10Gbps network switch and a dual 10Gbps network card. This is because SAGE transmits the video flows in raw media.

In case of an uncompressed Full HD video (1920x1080p, 24 fps and 24 bpp), the estimated throughput will be:

1920 * 1080 = 2073600 number of pixels
2073600 * 24 = 49766400 (24-bit color, 16.7M colors)
49766400 * 24 = 1194393600 bits /s (24 frames per second)
1194393600 bits /s ≅ **1.2 Gbps**

Backplane calculation

The Backplane capacity taking into account the previous example will be the following:

Backplane capacity = 1.2 Gbps + 1.2 Gbps = **2.4 Gbps**

Conclusion

The system implemented in this project has a throughput of 10 Mbps and a Backplane of 10 Mbps. The results obtained with SAGE are so far away from the results showed in this project because SAGE uses uncompressed video instead of using video compression. This, for sure, makes the difference in the cost of the elements and power consumption.

CHAPTER 11. PROJECT PLANNING

This chapter provides a summary of all the diferent tasks performed in this project and the time investment per task. Also, a time line is drawn to show the evolution of the project planning.

11.1. Task table

The following table shows a summary of all tasks performed in the project and the time investment to achieve them. It is necessary to say that 1 day is equivalent to 7 hours of work and the calculation is approximate.

PHASE	TASK	TIME (Days)
1. Initial Investigation and software installation	Initial investigation based on requirements	2
	Searching for information about VideoWalls	1
	Searching and analysing Raspberry Pi OS	2
	Installing Debian 7.1	1
	Installing Raspbian OS	3
	Installing Putty – Advanced IP Scanner	1
	Installing and test XBMC	1
	Investigating streaming frameworks	3
	Installing GStreamer (Server and Clients)	3
	Installing VLC (Server)	2
	Installing FFmpeg (Server)	1
	TOTAL	**20**
2. Study of the protocol architecture	Video compression techniques	3
	H.261 –H.263 – H.264 video codecs	4
	RTP and RTCP	1
	MPEG2-TS	3
	TOTAL	**11**
3. Development of Milestones	GStreamer Development	40
	FFmpeg Development	20
	VLC Development	10
	OMXPlayer development	10
	TOTAL	**80**
4. Synchronization	Investigation of GStreamer clocks	2
	SPS and PPS parameters	3
	Analysis of I, P and B frames	2
	Time drift calculations	1
	TOTAL	**8**
5. CPU Usage Analysis	GStreamer	1
	FFmpeg	1
	VLC	1
	OMXPlayer	1
	TOTAL	**4**
6. Network testing	Investigating DummyNet	2
	Designing scenario	1
	DummyNet configuration	2
	Performing test	3
	TOTAL	**8**

7. Throughput and Backplane	Measuring throughput with Wireshark	1
	Calculating Backplane	1
	TOTAL	**2**
8. Design of VideoWall control system	Investigating how to create startup script	2
	Defining control System	1
	Programming startup scripts	2
	TOTAL	**5**
9. Comparison with commercial products	Investigating SAGE	3
	Calculating the SAGE Budget	2
	Calculating the VideoWall budget	1
	Power comsumption analysis	1
	SAGE throughput and backplane	1
	TOTAL	**8**
10. Written Report	Writing the report	30
	Corrections	10
	TOTAL	**40**
11. Presentation	Creating the Power Point	7
	Preparing oral presentation	2
	TOTAL	**9**
TOTAL TIME INVESTMENT	**195 days**	

Table 14. Task table

11.2. Time line

A timeline is a way of displaying a list of events in chronological order, sometimes described as a project artifact.

Below these lines, the project time is shown:

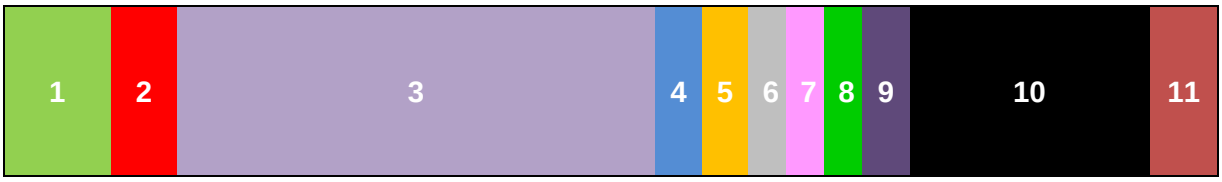


Figure 59. Project time line

Each colour shown in the time line corresponds to one of the phases listed in the table. The width of each phase is proportional to its time investment.

It is interesting to remark that phases 3 and 10 are the ones in which much time investment has been devoted to.

CHAPTER 12. CONCLUSIONS

The main milestone of this project was to build a cheap Video Wall using Raspberry Pi and we are proud to announce that this milestone is accomplished after working tirelessly during seven months.

It is important to remark that all the project milestones have been acomplished in the following way:

1. Low cost video wall is a reality. Using an average notebook acting as a Server and a set of Rasberry Pi devices, any person could build a cheap video wall, using free software as GStreamer and some time for installing it in the different end devices. Users would only need a typical ADSL router, UTP Cat 5e and HDMI cables to perform the video wall.

2. Raspberry Pi are powerful devices that can play back H.264 Full HD videos without any doubt.

3. The evidences shown in Chapter 5 demonstrate that the video wall implemented is synchronous.

4. We have learned everything related with H.264 encoding and transmission over the network using RTP as application layer and UDP as transport layer.

5. Minimum latency in reception. This point is very important. The value of latency that ensures a good performance in all the milestones defined in chapter 4 is 500 ms. A latency below this point does not guarantee a perfect visualization of Full HD videos. Obviously, if resolution is lower, less latency is needed to achieve a good visualization.

6. This project is scalable: the Raspberry Pi devices are in charge of cropping the input video flow and show it on screen. So, just by adding more Raspberry Pi running the developed scripts in this project the video wall could be extended for larger matrixes of monitors.

7. The different tests done with DummyNet demonstrates that the video wall is robust. It recovers perfectly when losses and delays are applied in the network.

Frameworks

In this section a brief summary of the milestones done during the project and the results obtained in each framework are provided.

These frameworks were new for us and we have learned how to use them: GStreamer, FFmpeg and VLC (with better and worse results). The final idea that we want to transmit is that building a low cost video wall is not easy, though it is possible, as it has been demonstrated in our project.

The framework analysis have been focused on GStreamer. GStreamer has demonstrated the ability to media-handling components. Engineers like us can take advantages of advances in codec and filter technology transparently and using different plugins available, users can build its own streaming application. Gstreamer provides hardware support in Raspberry Pi devices and also can be installed in the server.

FFmpeg is another insteresting framework able to decode, encode, transcode, mux, demux, stream, filter and play a lot of format. However, it does not provide hardware support of FFmpeg in the Raspberry Pi devices.

VLC, a portable free and open source media player and streaming media server, has compiled version with hardware acceleration. Although, it was impossible to install in our Raspberry Pi devices as commented in Chapter 4.

Finally OMXPlayer is a video player specifically made for the Raspberry Pi device but it was incompatible with the server OS.

The following table shows all the milestones done during the project and the final result for the 4 frameworks used: GStreamer, FFmpeg, VLC and OMXPlayer.

Objectives	GStreamer	FFmpeg	VLC	OMXPlayer
Playback H264 Full HD video	🟢	🔴	🔴	🟢
Send a HD video (local audio)	🟢	🟢	🟢	🔴
Send a Full HD video (local audio)	🟢	🟡	🟡	🔴
Send a real time Web Cam video flow	🟢			🔴
Receive a Full HD video / webcam flow and playback	🟢	🔴	🔴	
Send a HD video using MPEG-2 TS	🟢		🟡	🔴
Receive a HD video using MPEG2-TS	🟢	🔴	🔴	🟡
Send a Full HD video using MPEG-2 TS	🟢			🔴
Receive a Full HD video using MPEG2-TS	🟢	🔴	🔴	🟡
Send audio stream from a Full HD video	🟢			🔴
Receive Full HD audio stream	🟢	🔴	🔴	
Split and send a HD into two video flows	🟢	🟢		🔴
Split and send a Full HD into two video flows	🟢	🟡		🔴
Receive and crop Full HD video / webcam and playback	🟢	🔴	🔴	
Send a Full HD video (audio + video)	🟢			🔴

Table 15. Frameworks results comparison

We have defined 3 colour levels to make an easy evaluation of the results:

- The green circle 🟢 indicates that the results have been satisfactory.
- The yellow circle 🟡 means that the milestone has been accomplished but there are some problems with the video quality.
- The red circle 🔴 indicates that the milestone is not suitable for this framework (because it is not possible to install in our Raspberry Pi devices or in our Server).

The empty cells correspond to milestones that we have not focused on for this framework. As it can be observed, the Framework that obtains the best result is **GStreamer**.

Environmental impact

Finally, about the environmental impact of this project, it is important to take into account:

- Power consumption: A comparison with the commercial product SAGE has been done in Chapter 10, and the power consumption (Watts) of the Video Wall implemented is much lower. This fact reduces the environmental impact because it is possible to obtain more operating time for the same power.

- Components replacement: Due to the high grade of system modularity, it is easy, in case of failure, to replace just the broken component and keep working. Because Raspberry Pi are really small components, the material removed is much lower than other commercial cases like SAGE.

- VideoWall can be used for teleconferences. The advantages are the reduction of work trips and therefore a saving in terms of fuel consumed.

- Another advantage of the video wall is that having an array of screens, in case of failure, it is necessary to replace only one of them. In contrast, using a single large screen, if it breaks, it is necessary to replace the entire screen, with large environmental cost that entails.

CHAPTER 13. BIBLIOGRAPHY

Raspberry Pi

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ANNEXES

ANNEX A. RASPBIAN OS INSTALLATION

In this annex we will explain how to prepare a SD card for the Raspbian OS installation in our Raspberry Pi, perform the Raspbian OS installation and configure the Raspberry Pi network to allow internet connection.

A1. Preparing our SD card for the Raspberry Pi

In order to use our Raspberry Pi, we will need to install an Operating System (OS) onto a SD card. The following steps will allow us to install a recovery program onto a SD card that will allow us to easily install different OS's and to recover our card in case of accident:

1. Insert the 8GB SD card size into our computer.
2. Format the SD card so that the Pi can read it.
    a. Download the SD Association's Formatting Tool
    b. Install and run the Formatting Tool on our machine
    c. Set *"FORMAT SIZE ADJUSTMENT"* option to "*ON*" in the "*Options*" menu.
    d. Check that the SD card inserted matches the one selected by the Tool.
    e. Click the "Format" button.

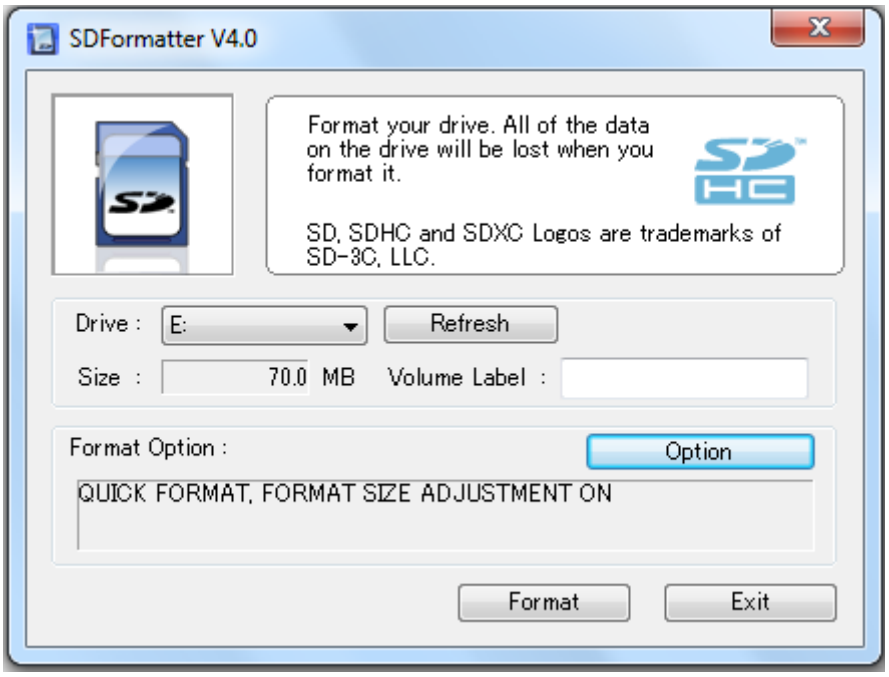


Figure A1. SD Formatter V4.0 Configuration

A2. Installation Raspbian “wheezy”

We have decided to install the Raspbian “wheezy” OS. This OS is based on the Raspbian optimized version of Debian, and containing LXDE, Midori, development tools and example source code for multimedia functions.

To perform the installation in Windows 7 we will need a program which is called *Win32DiskImager* and we will need to follow the next steps:

1. Download the distribution from the raspberrypi.org downloads page. We need to be sure that the distribution is for the Raspberry Pi, as others will not work. Usually these are zipped (compressed) files ending in .zip or .gz (something like "distribution-name.zip").

2. Extract the image file from the downloaded .zip file, so now we have "distribution-name.img".

3. Insert the SD card into our SD card reader and check what drive letter it was assigned.

4. Download the Win32DiskImager utility (it is also a zip file).

5. Extract the executable from the zip file and run the Win32DiskImager utility.

6. Select the image file we extracted above.

7. Select the drive letter of the SD card in the device box. We need to be careful selecting the correct drive; if we get the wrong one we could destroy our data on the computer's hard disk.

8. Click Write and wait for the write to complete.

9. Exit the imager and eject the SD card.

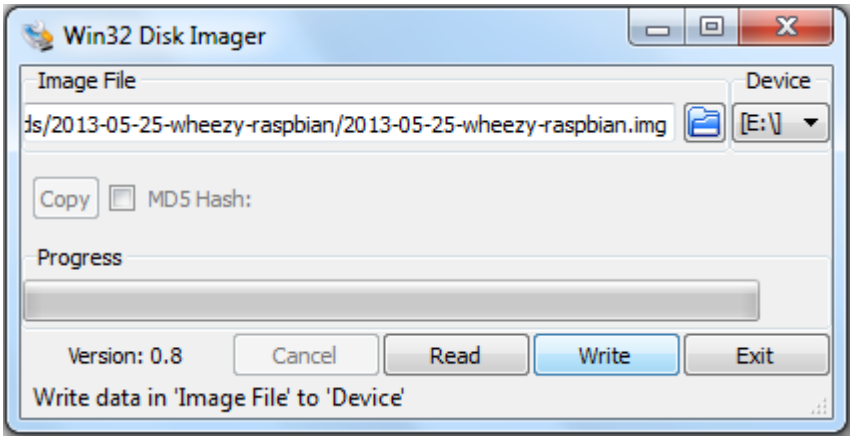


Figure A2. Win32 Disk Imager Configuration

10. We are now ready to plug the card into our Raspberry Pi.

A3. Configuration of Raspberry Pi Network

Initially we need to know the IP of our Raspberry. To archive that we will use a program which is called Advanced IP Scanner**.**

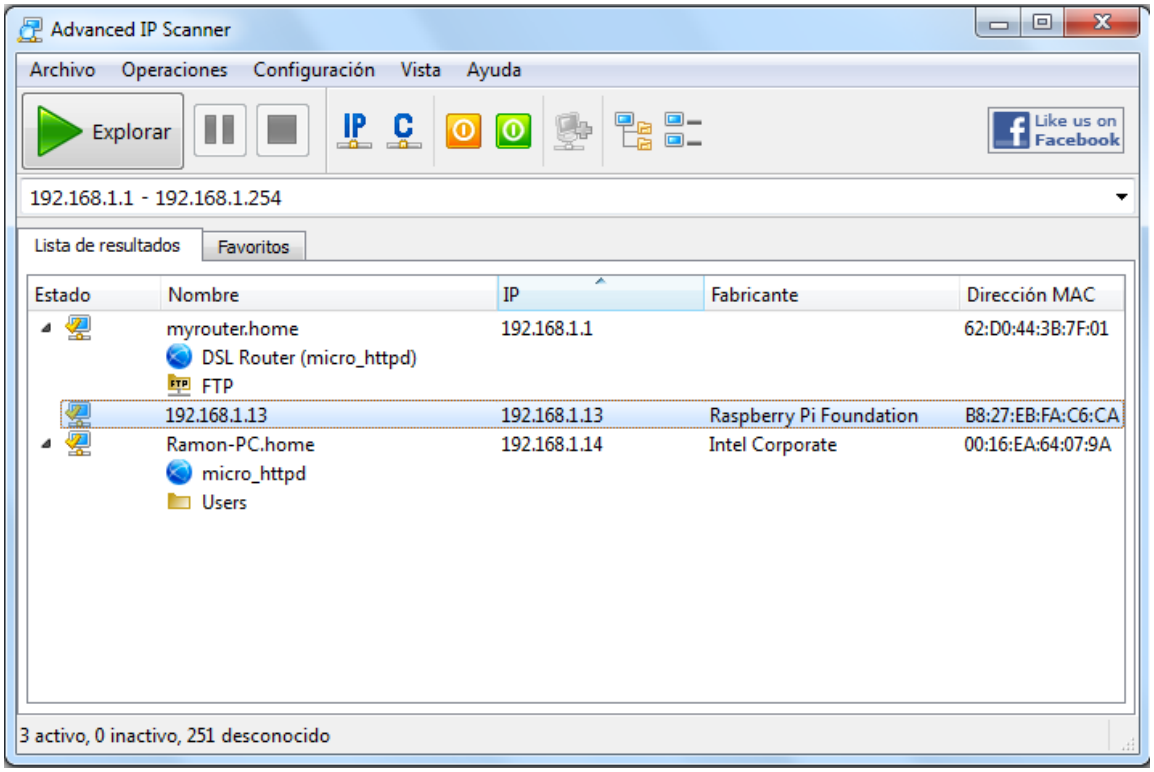


Figure A3. Advanced IP Scanner Program

To establish a connection between our PC and the Raspberry Pi we will need a program called PuTTY.

PuTTY is a free and open-source terminal emulator; serial console and network file transfer application. It supports several network protocols, including SCP, SSH, Telnet and rlogin. The name "PuTTY" has no definitive meaning, though *TTY* is the name for a terminal in the UNIX tradition, usually held to be short for Teletype.

First of all we need to define the basic options for the PuTTY session, this is Host Name (or IP address), Port and Type of connection.

Figure A4 and Figure A5 show our PuTTY configuration to establish connection through the Raspberry Pi.

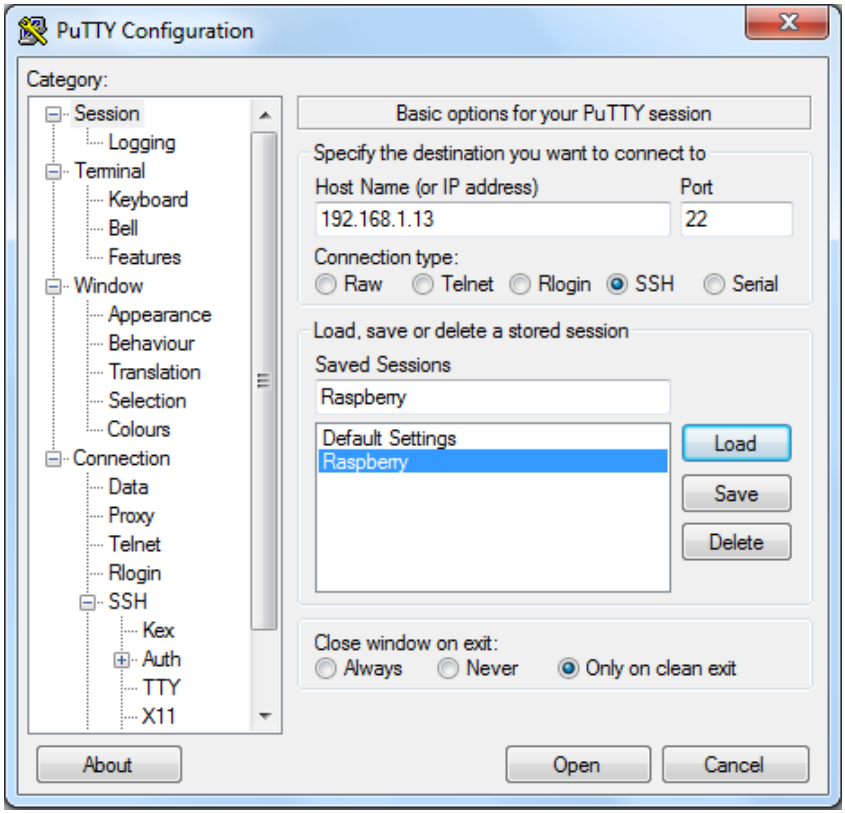


Figure A4. PuTTY Configuration (1)

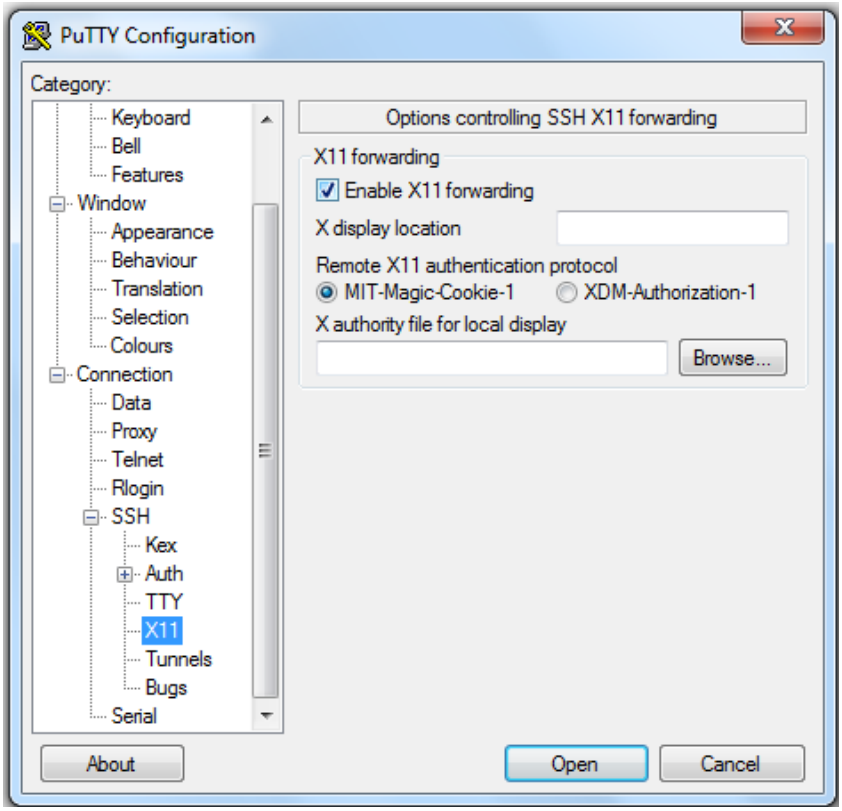


Figure A5. PuTTY Configuration (2)

Once we press Open button, the connection is established as we can see in Figure A6 and Figure A7.

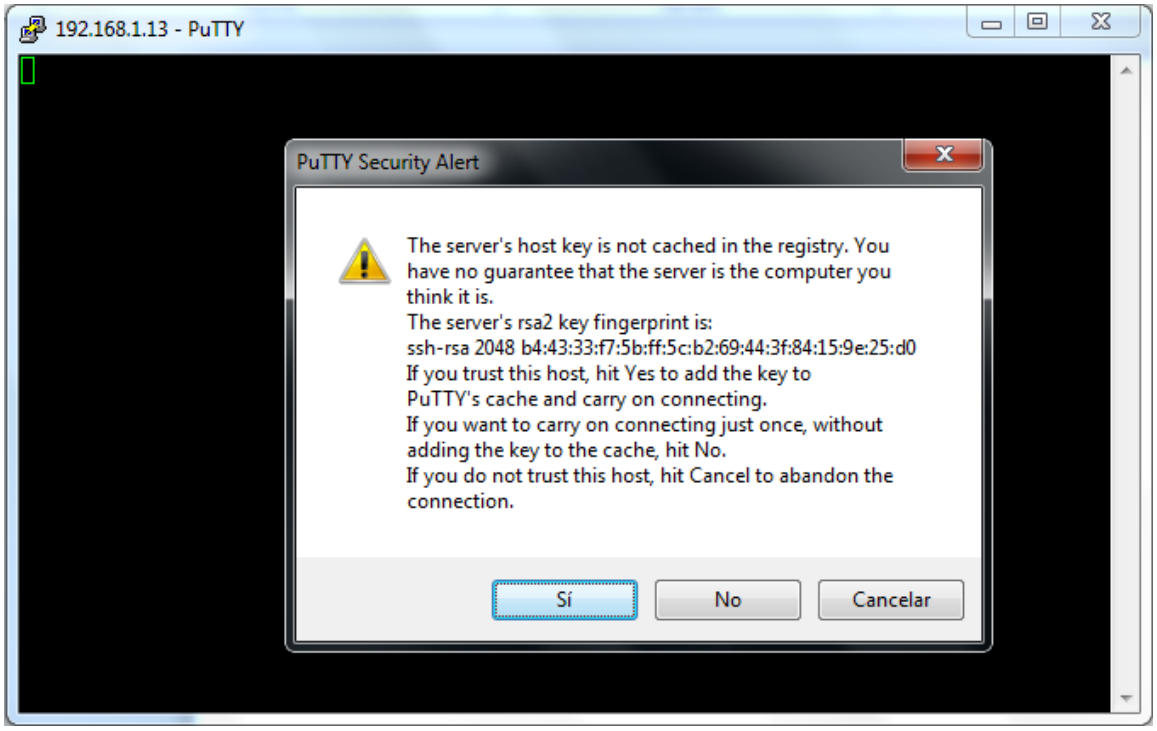

192.168.1.13 - PuTTY
PuTTY Security Alert
The server's host key is not cached in the registry. You have no guarantee that the server is the computer you think it is.
The server's rsa2 key fingerprint is:
ssh-rsa 2048 b4:43:33:f7:5b:ff:5c:b2:69:44:3f:84:15:9e:25:d0
If you trust this host, hit Yes to add the key to PuTTY's cache and carry on connecting.
If you want to carry on connecting just once, without adding the key to the cache, hit No.
If you do not trust this host, hit Cancel to abandon the connection.
Sí
No
Cancelar

Figure A6. Connection established

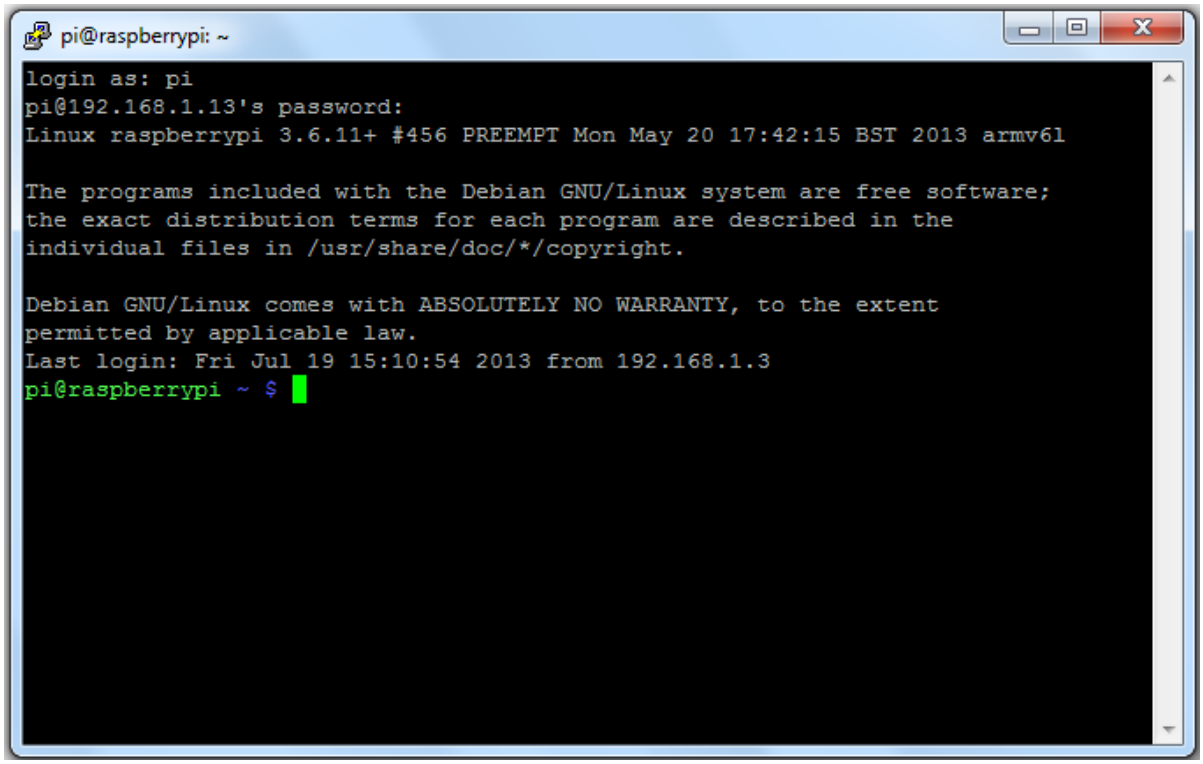

pi@raspberrypi: ~
login as: pi
pi@192.168.1.13's password:
Linux raspberrypi 3.6.11+ #456 PREEMPT Mon May 20 17:42:15 BST 2013 armv6l
The programs included with the Debian GNU/Linux system are free software;
the exact distribution terms for each program are described in the
individual files in /usr/share/doc/*/copyright.
Debian GNU/Linux comes with ABSOLUTELY NO WARRANTY, to the extent
permitted by applicable law.
Last login: Fri Jul 19 15:10:54 2013 from 192.168.1.3
pi@raspberrypi ~ $

Figure A7. Session terminal of Raspberry Pi

Next step is to share Internet to our Raspberry Pi. To make that possible we need to share the Wireless Connection with the Local Area Connection in Windows 7.

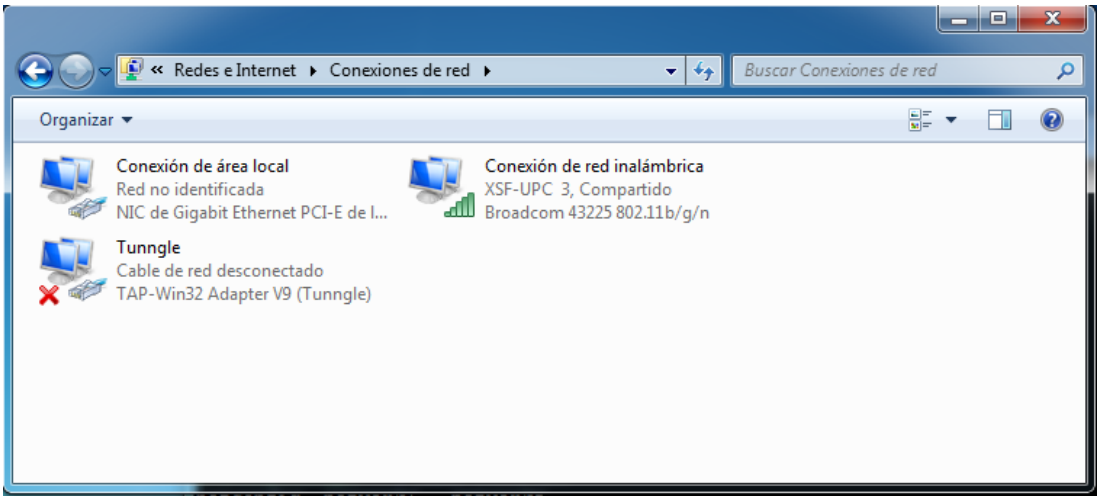


Figure A8.Wireless Connection shared - Windows 7

After that, we need to open the file */etc/network/interfaces* with a text editor (i.e. GNU Nano) to add a static address of our Raspberry Pi, a net mask and a gateway.

As it is shown in the Figure A9, the IP selected is 192.168.137.3 /24 because the Ethernet interface of our PC has the IP 192.168.137.1 /24.The gateway of the Raspberry Pi is the Ethernet interface of our PC.

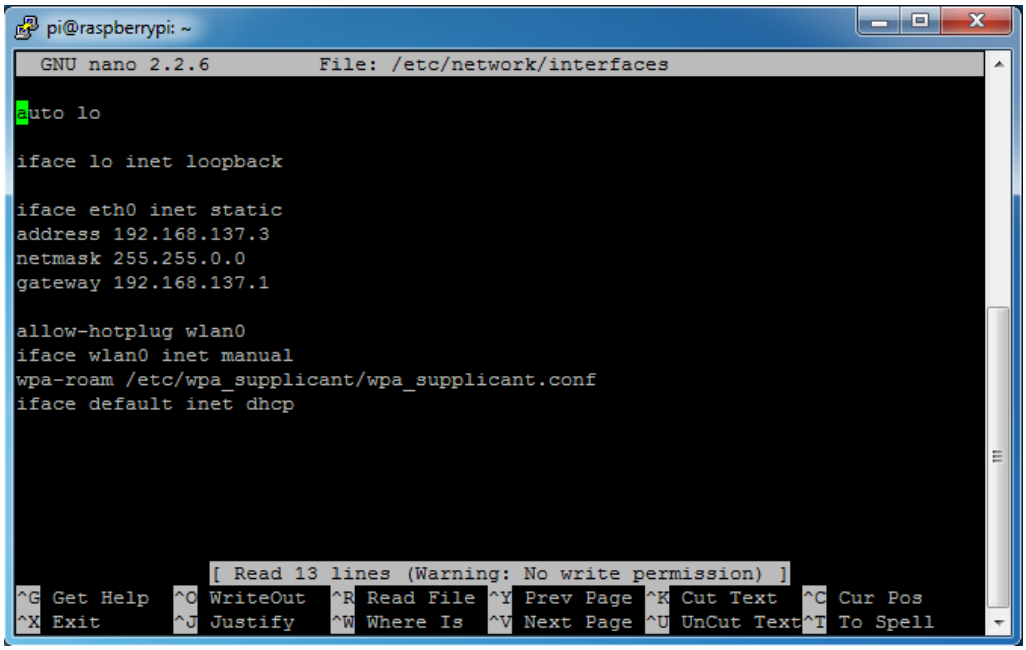

pi@raspberrypi: ~
GNU nano 2.2.6 File: /etc/network/interfaces
auto lo
iface lo inet loopback
iface eth0 inet static
address 192.168.137.3
netmask 255.255.0.0
gateway 192.168.137.1
allow-hotplug wlan0
iface wlan0 inet manual
wpa-roam /etc/wpa_supplicant/wpa_supplicant.conf
iface default inet dhcp
[Read 13 lines (Warning: No write permission)]
^G Get Help ^O WriteOut ^R Read File ^Y Prev Page ^K Cut Text ^C Cur Pos
^X Exit ^J Justify ^W Where Is ^V Next Page ^U UnCut Text ^T To Spell

Figure A9. Network interface configuration

In addition it is necessary to configure the DNS. We need to open the file */etc/resolv.conf* and add a *nameserver (*in this case, the name server is 147.83.2.3 which is the DNS of UPC). The other two name servers are the ones that we use in our home, so Raspberry Pi can work in the different wireless networks that we will use during this project.

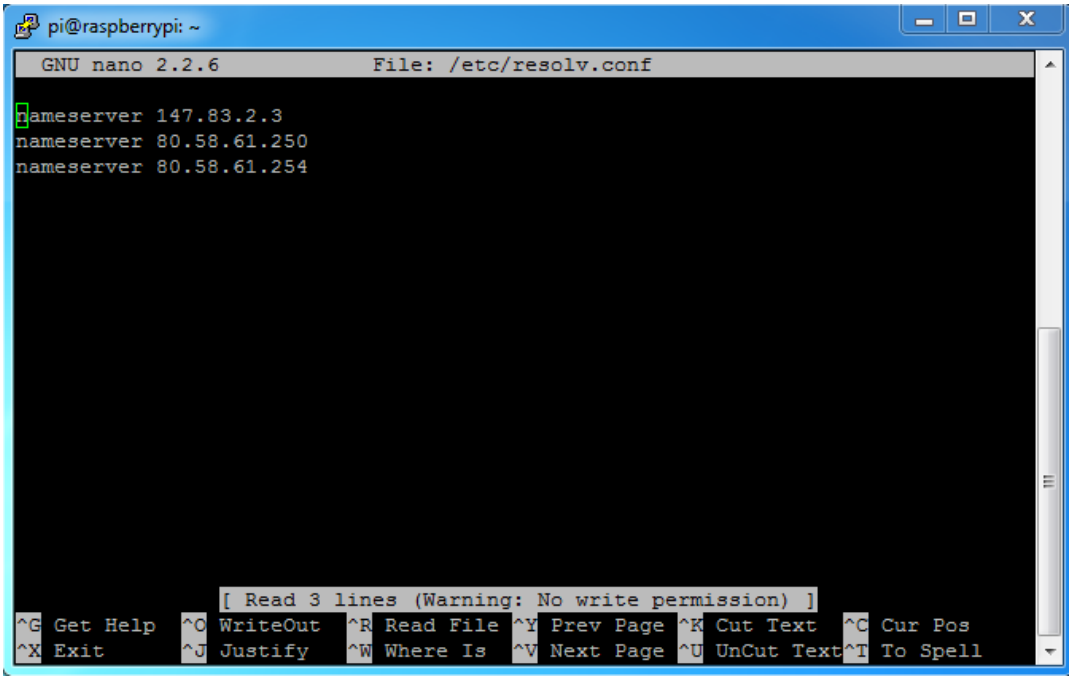

pi@raspberrypi: ~
GNU nano 2.2.6 File: /etc/resolv.conf
nameserver 147.83.2.3
nameserver 80.58.61.250
nameserver 80.58.61.254
[Read 3 lines (Warning: No write permission)]
^G Get Help ^O WriteOut ^R Read File ^Y Prev Page ^K Cut Text ^C Cur Pos
^X Exit ^J Justify ^W Where Is ^V Next Page ^U UnCut Text ^T To Spell

Figure A10. DNS configuration file

ANNEX B. GSTREAMER INSTALLATION

By default GStreamer 0.10 is installed in the Raspbian OS. But, due to project requirement a GStreamer 1.0 installation will be performed in both Server and Clients.

B1. Installing GStreamer 1.0 in the Server

Due there is no an official package of GStreamer 1.0 available, we need to compile and install GStreamer manually. As you will see next, there are a set of different plugins that should be installed to work with GStreamer and they are classified as base, good, bad and ugly plugins. Base and good plugins work perfect with GStreamer but bad and ugly plugins are not working correctly sometimes.

```
$ sudo apt-get install libxv1 libxv-dev libxvidcore4 libxvidcore-dev
faad libfaad-dev bison  libavl-dev yasm flex  zlib1g-dev  libffi-dev
                              gettext
```

Install the latest version of glib (2.32 is required)

```
$ wget http://ftp.gnome.org/pub/gnome/sources/glib/2.34/glib-
2.34.1.tar.xz
$ tar xvf glib-2.34.1.tar.xz
$ cd glib-2.34.1
$ ./configure --prefix=/usr
$ sudo apt-get install make
$ make
$ sudo make install
```

Install the packages and plugins

```
$ wget http://gstreamer.freedesktop.org/src/gstreamer/gstreamer-
1.0.2.tar.xz
$ tar xvf gstreamer-1.0.2.tar.xz
$ cd gstreamer-1.0.2
$ ./configure --prefix=/usr
$ make
$ sudo make install
```

Install coders and stuff

```
$ sudo apt-get install libtheora-dev libogg-dev libvorbis-dev
libasound2-dev libjack-dev
```

Install libvisual-0.4-dev for xvimagesink (if running X11)

```
$ sudo apt-get install libxv-dev libvisual-0.4-dev
```

Install base plugins

Gstreamer base plugins is a well-groomed and well-maintained collection of GStreamer plugins and elements. Also contains helper libraries and base clases useful for writing elements. A wide range of video and audio decoders, encoders, and filters are included.

```
$ wget  http://gstreamer.freedesktop.org/src/gst-plugins-base/gst-plugins-base-1.0.2.tar.xz
$ tar xvf gst-plugins-base-1.0.2.tar.xz
$ cd gst-plugins-base-1.0.2
$ ./configure --prefix=/usr
$ make
$ sudo make install
```

Install the good plugins

This is a set of plugins that Gstreamer team considers to have good quality code, correct functionality and LGPL license.

```
$ wget  http://gstreamer.freedesktop.org/src/gst-plugins-good/gst-plugins-good-1.0.2.tar.xz
$ tar xvf gst-plugins-good-1.0.2.tar.xz
$ cd gst-plugins-good-1.0.2
I did for some reason have problem making the goom filter and did therefor exclude it with the --disable-goom
$ ./configure --prefix=/usr --disable-goom
$ make
$ sudo make install
```

Installing the RTMP library

```
$ sudo apt-get install librtmp-dev
```

Install the bad plugins

This is a set of plugins that are not up to par compared to the rest. They might be to close to being good quality, but are missing something – be it a good code review, some documentation, a set of tests, a real live maintainer o some actual wide use.

```
$ wget http://gstreamer.freedesktop.org/src/gst-plugins-bad/gst-plugins-bad-1.0.2.tar.xz
$ tar xvf gst-plugins-bad-1.0.2.tar.xz
$ cd gst-plugins-bad-1.0.2
$ ./configure --prefix=/usr
$ make
$ sudo make install
```

Install the ugly plugins

This is a set of plugins that have good quality and correct functionality, but distributing them might pose problems. The code might be widely known to present patent problems.

```
$ sudo apt-get install libmad0-dev libx264-dev
$ wget http://gstreamer.freedesktop.org/src/gst-plugins-ugly/gst-plugins-ugly-1.0.2.tar.xz
$ tar xvf gst-plugins-ugly-1.0.2.tar.xz
$ cd gst-plugins-ugly-1.0.2
$ ./configure --prefix=/usr
$ make
$ sudo make install
```

B2. Installing GStreamer 1.0 in the Clients

The installation in the Raspberry Pi is simple. Just adding a source list and then just automatically install the package.

```
nano /etc/apt/sources.list
deb http://vontaene.de/raspbian-updates/ . main
sudo apt-get install gstreamer1.0
```

ANNEX C. VLC Installation

C1. Installing VLC in the Server

It is really simple to install VLC in the server. It is highly recommended to get a recent version of VLC on wheezy by installing the version from **wheezy-backports**.

We need to add the wheezy-backports repository to our **sources.list** by writing:

```
deb http://mirror.debian.org/debian wheezy-backports main
```

(or add a new file with the ".list" extension to /etc/apt/sources.list.d/)

Next, we need to run **apt-get update.**

To install VLC:

```
sudo apt-get -t wheezy-backports install vlc
```

Without backports, we will end up with the very old version of VLC in vanilla wheezy (install vlc and browser-plugin-vlc as usual).

C2. Installing VLC in the Clients

In October 2013, a VLC version with hardware acceleration enabled on Raspberry Pi was compiled using Raspbian.

We were interested on that and we downloaded the compiled version. We followed next the steps provided in different web pages:

```
Bootstrapping VLC
=================
Please run: ./bootstrap

Configuring VLC
===============
Please run: ./configure

The recommendation is to use GCC to build VLC.

Building VLC
============
Once configured, run `make' to build VLC.

Installing and running VLC
==========================
Install the VLC and its plugins by typing: make install

Building packages
=================

To build a Debian package, you need to get the packaging info git clone
git://git.debian.org/pkg-multimedia/vlc.git debian and then git-buildpackage.
```

After follow these steps, different errors appeared during the installation that were impossible to solve for us and thus we decided to not install VLC on the Raspberry Pi devices.

An alternative method that it is expensive in time (around 20 hours, and that time investment does not guarantee a success) is to compile ourselves the VLC.

The instructions to compile it by ourselves are the following:

Step 0: environment preparation - build tools installation

```
# apt-get install git libtool build-essential pkg-config autoconf
```

Step 1: get VLC sources from git

```
# git clone git://git.videolan.org/vlc.git
# cd vlc
# ./bootstrap
```

Step 2: Configure enabling the important modules for hardware acceleration on raspberry.

install all these libraries needed by VLC and it's modules:

```
liba52-0.7.4-dev libdirac-dev libdvdread-dev libkate-dev
libass-dev libbluray-dev libcddb2-dev libdca-dev
libfaad-dev libflac-dev libmad0-dev libmodplug-dev
libmpcdec-dev libmpeg2-4-dev libogg-dev libopencv-dev
libpostproc-dev libshout3-dev libspeex-dev libspeexdsp-dev
libssh2-1-dev liblua5.1-0-dev libopus-dev libschroedinger-dev
libsmbclient-dev libtwolame-dev libx264-dev libxcb-composite0-dev
libxcb-randr0-dev libxcb-xv0-dev libzvbi-dev
```

```
# ./configure --enable-rpi-omxil --enable-dvbpsi
--enable-x264 <other_modules_you_might_need>
```

The modules listed above used to configure the installation are the ones absolutely needed to enable hardware acceleration. It might need to enable or disable some others depending on the needs.

Depending on the modules enabled it will be needed to install third party libraries.Configure will fail and tell what library is missing. In this case it has to install it and run configure all over again. Bring patience, may be it has to repeat this a few times.

Step 3: Compile

```
# make clean
# make
```

Step 4: how to run it with hardware acceleration enabled from command line

```
# ./vlc --vout omxil_vout <other_params_you_might_need>
```

It is absolutely necessary to use the parameter **--vout omxil_vout** for VLC to use the custom video output for Raspberry Pi device.

Step 5 (optional)

To play back HD content it is strongly recommended to overclock the Raspberry. The file to change CPU frequency is /boot/config.txt.

We followed the 6 steps commented above, but due to the complexity and the constant request of multiple third party libraries, this option has a very low probability of success. We devoted 2 days (7 hours per day) but the compilation failed.

ANNEX D. H261 and H263 video codecs

D1. H.261 video codec

H.261 is a video coding standard published by ITU in 1990. It is designed for ISDN network data rates which are multiples of 64 kbit/s. A functional block diagram of the video codec is shown in Figure B1.

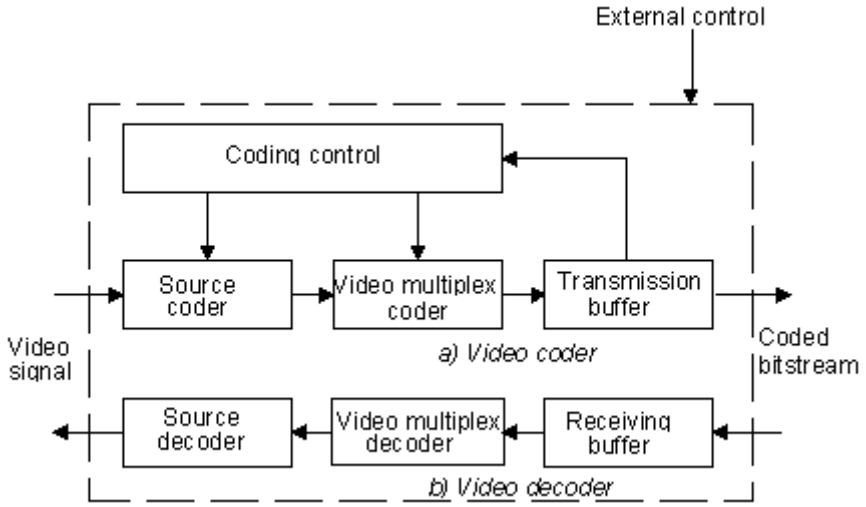


Figure B1. A block diagram of H.261 video codec

The main element in the encoding process is a source coder which compresses the incoming video signal. The source coder operates on non-interlaced pictures occurring 30000/1001 times per second. Pictures are coded as values of the luminance and two colour difference components (Y, Cb, Cr). These components are defined in ITU-R BT.601 recommendation. The Cb and Cr matrices are a quarter the size of the Y matrix. To solve the incompatibility between different TV standards (PAL, SECAM and NTSC), the CIF and QCIF picture structures were adopted. The QCIF format, which employs half the CIF spatial resolution, must be supported by all H.261 compatible codecs. QCIF is intended for videophone applications where head-and-shoulders pictures are sent. The full CIF format is used when several people must be viewed in a conference room, for example.

Picture Format.	Lumin. pixels	Lumin. lines	Use
QCIF	176	144	Mandatory
CIF	352	288	Optional

Table 16. H261 picture format

Each GOB is divided into 33 macroblocks. The macroblock header defines location of a macroblock within the GOB, type of coding, possible motion vectors, and which blocks will actually be coded within the macroblock.

Macroblocks in GOB										
1	2	3	4	5	6	7	8	9	10	11
12	13	14	15	16	17	18	19	20	21	22
23	24	25	26	27	28	29	30	31	32	33

Table 17. Macroblocks in GOB - H261

Each macroblock is further divided into six blocks. Four of them represent the luminance and two the chrominance components of the signal. Each block has 8x8 pixels, so the color resolution is half of the luminance resolution in both dimensions.

<table>
<tr><td colspan="2">Luminance (Y)</td><td>Chrominance / blue (Cb)</td><td>Chrominance / red (Cr)</td></tr>
<tr><td>1</td><td>2</td><td rowspan="2">5</td><td rowspan="2">6</td></tr>
<tr><td>3</td><td>4</td></tr>
</table>

Table 18. Luminance and Chrominance per macroblock

Above there is a short description of the coder's main functions:

Prediction

The prediction is inter-picture and may be augmented by motion compensation. The coding mode in which prediction is applied is called "inter". The coding mode is called "intra" if no prediction is applied. The intra coding mode comprise the picture level, and blocks (8x8 pixels) are encoded only with reference to themselves and are sent directly to the block coding process.

The inter coding uses another picture frame as a reference. The prediction error is calculated between corresponding macroblocks in the current and previous frames. A criteria if a block is transmitted or not, is not defined in the standard.

Motion compensation

Motion compensation support in a H.261 encoder is optional, but it is required from the decoder. In motion compensation the previous frame is searched for the best reference macroblock. The prediction error and motion vectors are sent to the block transformation process. Neither the search area nor the method to compute motion vectors are included in the standard.

Block coding

In block coding both intra coded and inter coded frames are coded as 8x8 blocks using two dimensional DCT, which gives an 8x8 coefficient matrix for each block.

Quantization

After DCT the coefficients are quantized. This step produces coefficients values, which are much smaller in magnitude than the original values and most of them become zero. The purpose of this step is to represent the coefficients with no greater precision than is necessary to achieve the required image quality.

Entropy coding

Extra lossless compression is achieved by RLE-coding coefficients and then further entropy coding these with Huffman-technique. This produces a variable length code.

Multiplexing

The video multiplex coder combines the compressed data with various side information, which indicates various operation modes, into hierarchical bitstream that can be interpreted by decoders.

Transmission buffer

The transmission buffer is employed to smooth the varying bit rate from the source coder to adapt it to the communication channel.

The coded bitstream is composed of two types of frames: intra coded I-frames and inter coded P-frames. An example of an encoded sequence might be:
I P P P I P P P .... where there are three predicted P-frames after each I-frame.

D2. H.263 video codec

The ITU-T H.263 recommendation is based on H.261 and is targeted for compressing the moving picture at low bitrates.

The coding algorithm of H.263 has some improvements and changes over that used by H.261 and it can often achieve the same quality as H.261 with less than half the number of bits in the coded stream. That is why it is replacing H.261 in many applications in the Internet, where bandwidth is often a critical resource.

H.263 codec has the following new features over H.261:

Half pixel precision is used for the motion compensation, as opposed to H.261 where full pixel precision and loopfilter are used.

Some parts of the datastream structure are now optional, so the codec can be configured for a lower bitrate or better error recovery.

There are four negotiable coding options included:Unrestricted motion vectors mode, syntax-based arithmetic coding, advanced prediction mode and PB-frames mode.

The H.263 supports new picture resolutions.

Picture Format	Lumin. Pixels	Lumin. Lines	Use
SQCIF	128	96	Mandatory
QCIF	176	144	Mandatory
CIF	352	288	Optional
4CIF	704	576	Optional
16CIF	1408	1152	Optional

Table 19. H263 picture formats

The PB-frame consists of two frames coded as one unit. The P-frame is predicted from the previous P-frame and the B-frame which is predicted from both the previous and the current P-frame.

ANNEX E. GSTREAMER PIPELINES GRAPHS

To create a useful diagram of our pipelines we have followed the next steps:

1. Install dot

```
sudo apt-get install graphviz
```

The "*dot*" command is available in the "*graphviz*" package.

2. Run in the prompt:

```
# export GST_DEBUG_DUMP_DOT_DIR=/tmp/
```

This will place the .dot generated files in the /tmp/ directory. We can change that directory if we need to.

3. Run the pipeline

```
gst-launch-1.0 filesrc location= /home/carlos/Descargas/
Monsters.mp4 ! qtdemux name=dmux dmux.video_0 ! queue !
decodebin ! autovideosink dmux.audio_0 ! queue ! aacparse ! faad
! alsasink
```

After the pipeline is over, we can see the .dot generated files, and the "**PLAYING_PAUSED**" one is typically used to generate the diagram.

```
# ls /tmp
 0.00.00.972540004-gst-launch.NULL_READY.dot
 0.00.01.051387461-gst-launch.READY_PAUSED.dot
 0.00.01.074729712-gst-launch.PAUSED_PLAYING.dot
 0.00.59.857852589-gst-launch.PLAYING_PAUSED.dot
 0.00.59.961485839-gst-launch.PAUSED_READY.dot
```

4. Generate the diagram from the dot file

We can generate the .png image using the "*dot*" command:

```
$ dot -Tpng 0.00.59.857852589-gst-launch.PLAYING_PAUSED.dot >
pipeline.png
```

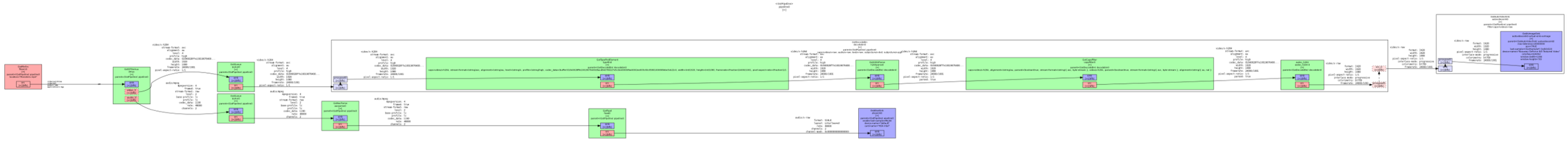
5. Open up your image

Once the image has been created, we can open it with our favourite image viewer, for example:

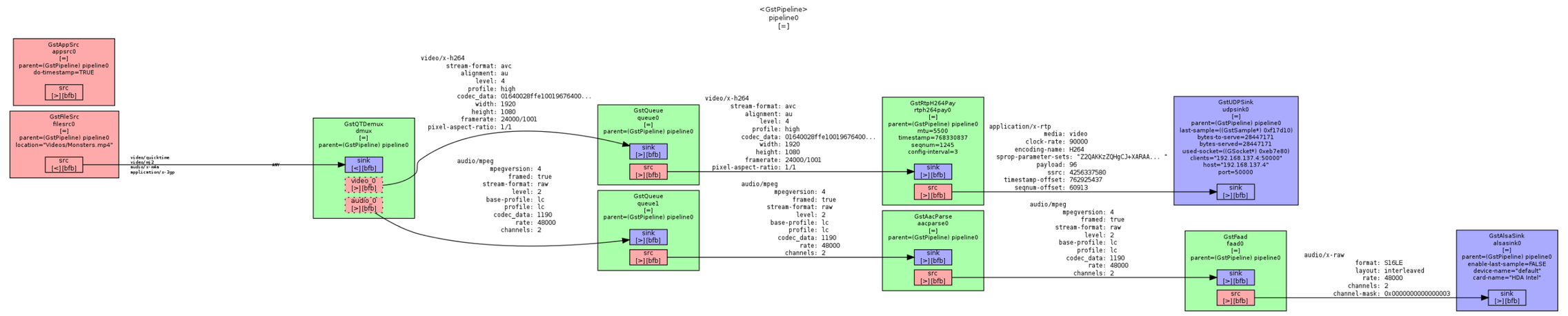
```
$ eog pipeline.png
```

The pipeline representation is shown in the following pages:

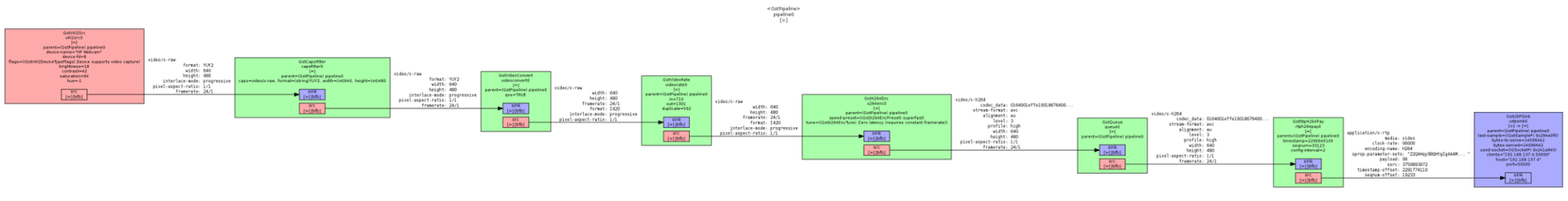
E1. Milestone 1



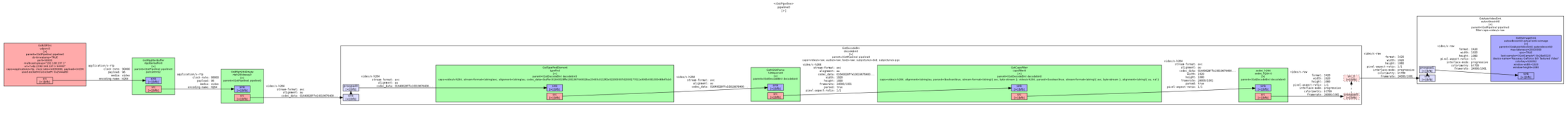
E2. Milestone 2



E3. Milestone 3

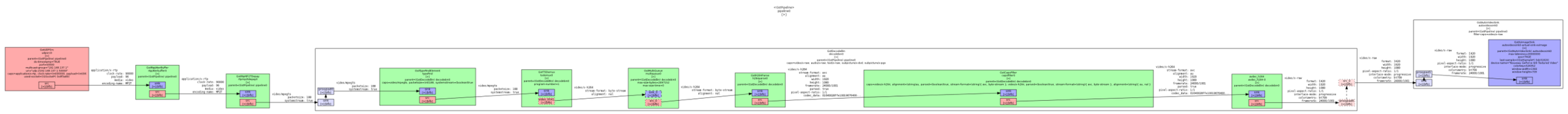


E4. Milestone 4



E5. Milestone 5

E6. Milestone 6



E7. Milestone 9